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Market Update

Bone Conduction Transducers

By Mike Klasco and J. Martins

Practical Test & Measurement

The Benefits of Using Structure-Borne Sound in Headset Signal Processing

By Hans W. Gierlich

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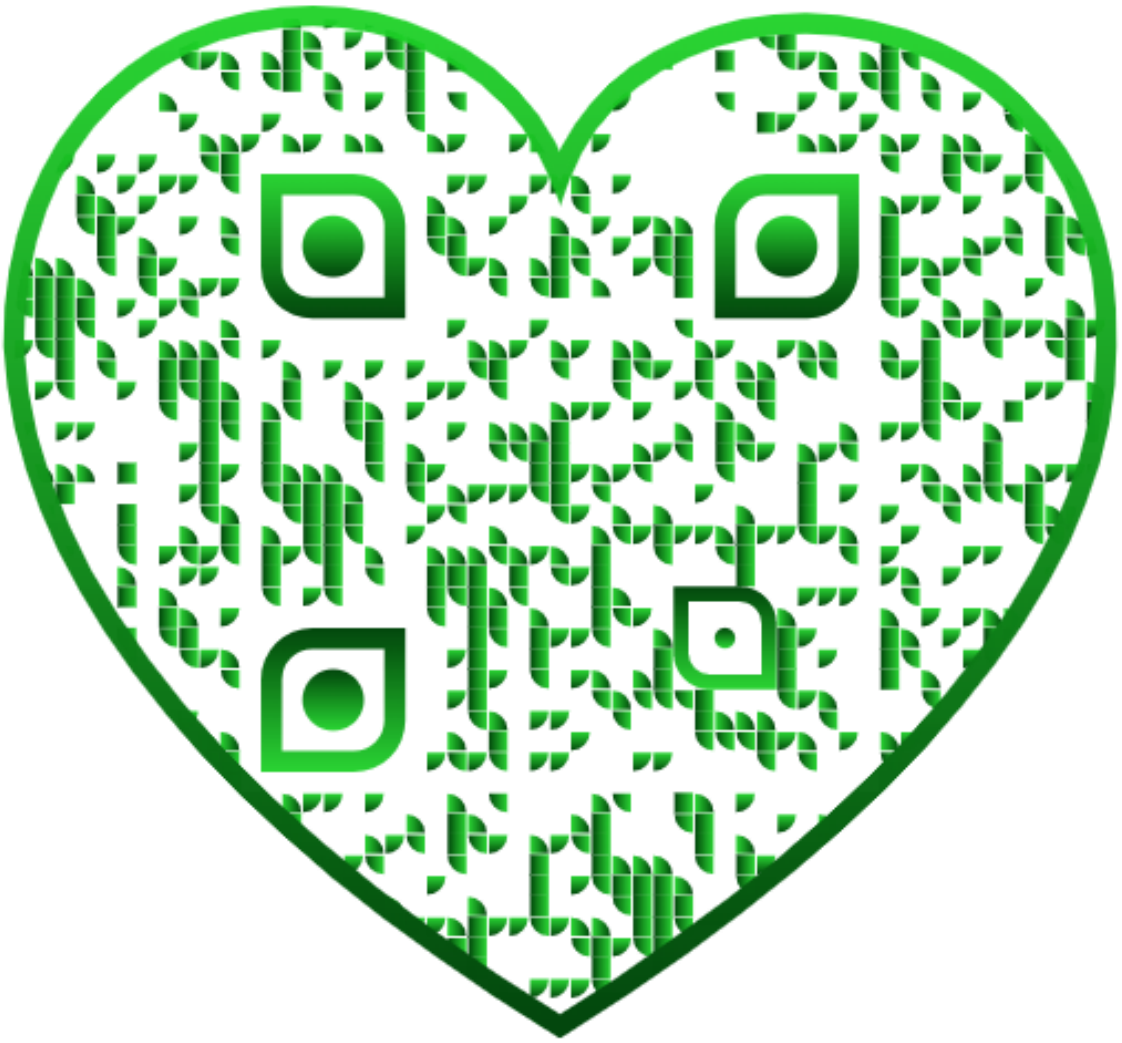
JANUARY 2023



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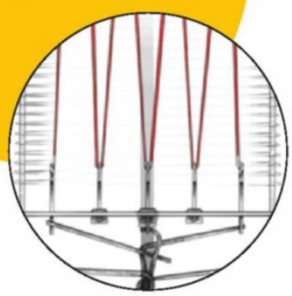

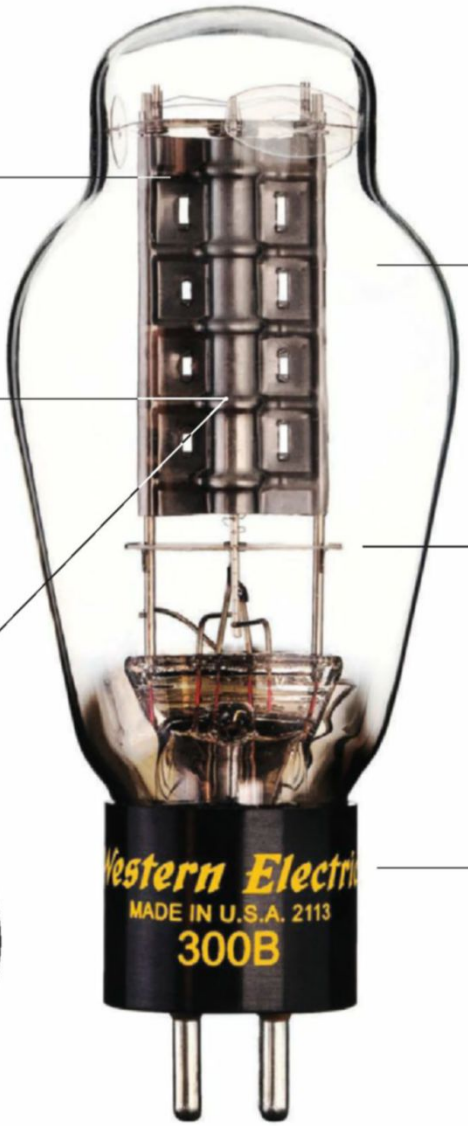

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From Ideas to Manufacturing

If you are holding a printed copy of this issue of *audioXpress* or reading it on a screen, you have probably already noticed that this publication is unlike anything else out there in the audio industry. This publication was originally born from what was considered once a very niche market segment of people who preferred a hands-on approach to audio systems and reproduction, and liked to modify and (hopefully) improve existing equipment, as well as design and build their own gear from scratch.

That spirit inspired the late Edward T. Dell, Jr. (1923–2013) to devote decades of his life to the DIY audio passion, leading to the creation of hallmark publications including *Audio Amateur* (1970), *Audio Electronics*, *Glass Audio*, *Speaker Builder*, *Multimedia Manufacturer*, and finally *audioXpress*, starting in January 2001. What many people recognized was that their passion for audio electronics and speaker building, among other things, fostered many ideas that led to the foundation of new companies, business careers, and audio innovation. Many of the original *audioXpress* readers were actually active members of the industry and others got involved because of it. All over the world, the pages of Ed Dell's publications and the valuable article contributions by many hundreds of experts, inspired new ventures globally.

I joined *audioXpress* as editor-in-chief in 2013, and I was one of those people who these magazines had touched, so when invited I accepted the challenge of leading this publication with the inspiration and conviction that there was a lot more that we could do to serve our audio product development and design readership.

We are now entering a new expansion phase in the trajectory that was initiated in 2013, when we embraced our mission to follow audio innovation, true to our motto, Advancing the Evolution of Audio Technology. And we continue to expand on our mission to offer the leading Audio Development and Design Magazine and Website. For that, you will note the important contributions from new authors and audio experts in almost every issue. This January 2023 issue is no different and I am particularly glad to welcome Andrew Bellavia and René Christensen, who I hope will be regular contributors to this publication. When you read their articles, I am certain you will agree.

A fundamental contributor to this project, as many of our authors recognize, has been the work of Jan Didden, who for many years has been listed as our Technical Editor, although his involvement was mostly voluntary. As we aim to meet the evolving needs of our audience, I reached out to him to become more closely involved with the publication, essentially expanding on what he always did brilliantly.

I am extremely pleased to announce that Jan Didden has agreed to take on the formal duties as Technical Editor of *audioXpress* (we decided that there wasn't a better title for what he does), expanding his involvement and contributions to this publication. And appropriately, I leave you with a word from him.

J. Martins
Editor-in-Chief

It is a privilege for me to be part of the *audioXpress* team. I cut my writing teeth many, many years ago producing articles for *The Audio Amateur*, the precursor to this title, founded by Ed Dell. Then, in 2010 I struck out on my own and started *Linear Audio*, publishing a 200-page "bookzine" with some 10 technical audio articles every six months. *Linear Audio*'s author roster grew to more than 100; experienced designers, well-known names in audio, as well as accomplished amateurs. You know who you are. In 2017, I decided I wanted to spend more time on research and equipment design and stopped publishing new *Linear Audio* issues. Occasionally I hooked up a prospective author with *audioXpress*, and this activity will now expand.

I believe that design and/or construction articles are an indispensable part of a technical audio journal. It is important to know what is out there, what's available, who offers what, how it performs, and how it tests. But at the start of it all, there is an engineer or designer who creates the equipment that is proposed, tested, offered, and advertised. I want to give this part of the community the visibility it deserves, to inform and educate and hopefully inspire technically inclined persons to take up audio design. I am looking forward to once again interacting with established and budding authors and designers to explore this great vocation of ours! So, get in touch with me at techeditor@audioXpress.com.

Jan Didden
Technical Editor

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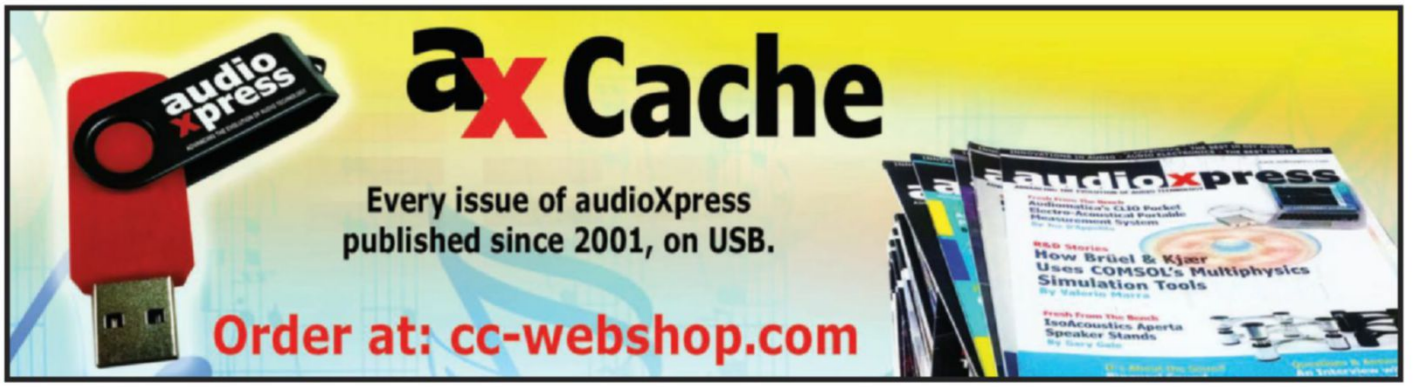

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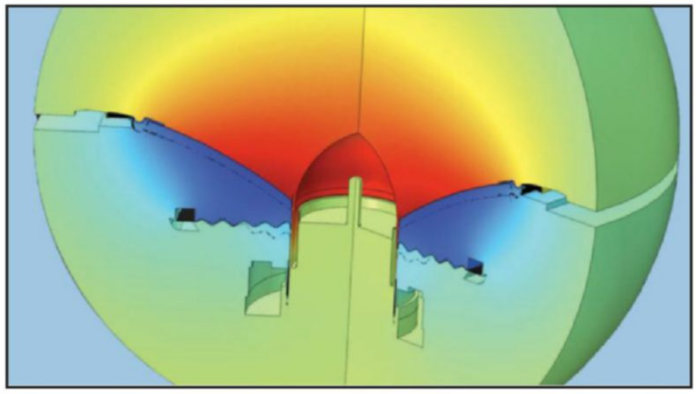

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Haylou PurFree Bone Conduction Earphones

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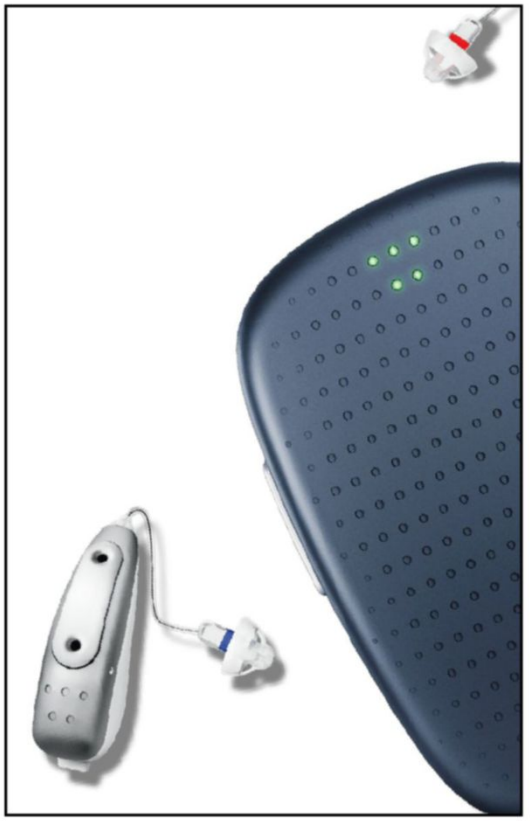

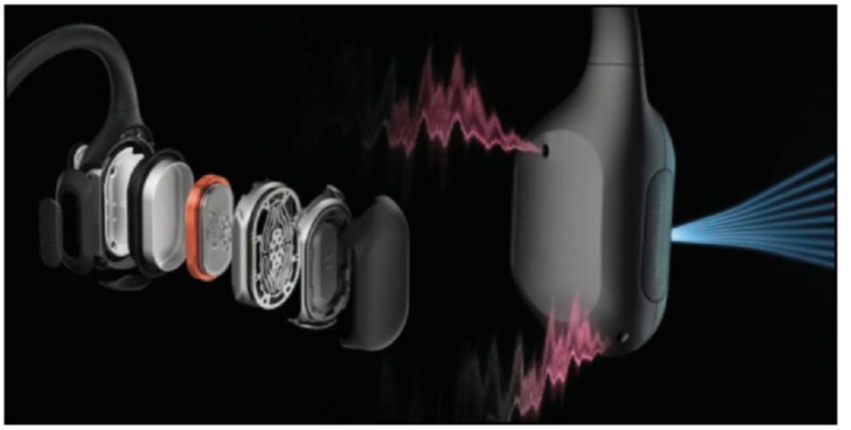

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Bone Conductors Going Mainstream

By
Mike Klasco and
J. Martins

Bone conduction technology that sends vibrations directly to the inner ear, bypassing the eardrums, is seeing increasing use—evolving from its origins in medical hearing devices to mainstream consumer and strategic communication applications. The advantages in open-ear wireless applications have great appeal and the audio industry is responding.

Popular dynamic headphone drivers, the most critical component and the voice of the headphone, are mostly commodity stuff and all are too close a cousin to the buzzer and microspeakers from which they descended. This should not be a surprise as many Asian headphone manufacturers started out producing buzzers and injection-molded plastic parts.

Conventional earphones do the job nicely for telephony communications and music listening for most consumer applications. The drivers on these designs produce sound by vibrating the air that reaches the external ear, eardrum, middle ear, and finally the inner ear. Yet, in-ear models are uncomfortable for many people for quite a few reasons. We all have experienced the pressure and lack of ventilation in our ear canals, the occlusion effect of standing waves when our ears are plugged up, along with the loss of situational awareness. Loose-fit earbuds have changed the earphone market forever, as demonstrated by the massive success of Apple's original AirPods.

Leaving the ear totally open by design, bone conduction "headphones" typically transmit vibrations directly to the human skull and skin, bypassing the ear canal, straight into the inner ear. The key value of bone conduction is that our ears are left completely open, allowing us to have a parallel hearing channel to our surroundings, while still enjoying the music. But to get to designs that are able to rival traditional earphones in music fidelity, we still have some way to go. Yet the key proposition in bone conduction is already appealing to a significant number of consumers and professionals.

Sound Through Our Bones

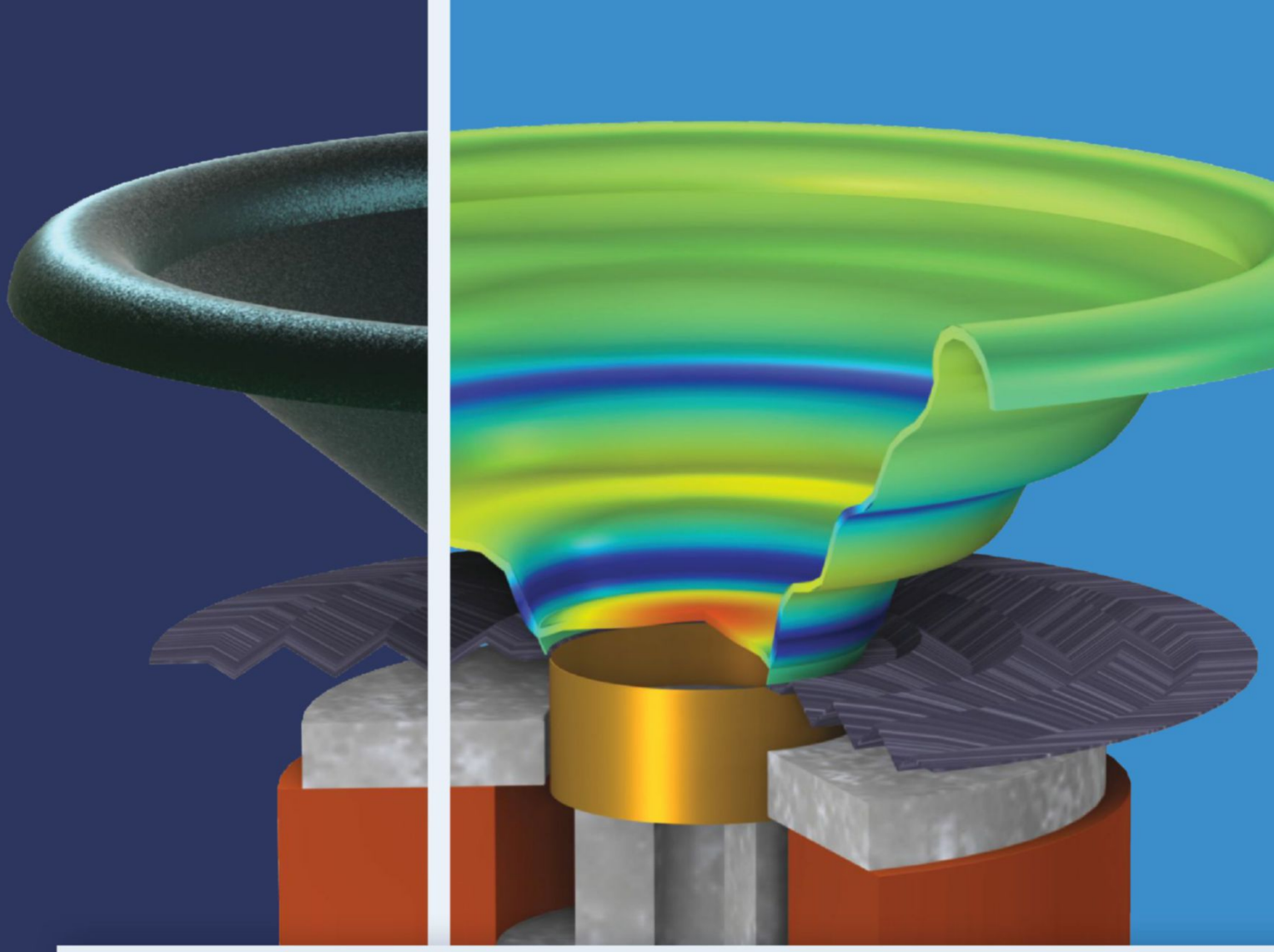
Long before the electroacoustic transducer was invented, bone conduction was already understood. In the 15th century Italian physician and mathematician Girolamo Cardano, a prolific inventor, described the fact that deaf people could "hear" music by connecting a rod attached to a musical instrument between someone's teeth. And Beethoven is said to have used that technique (putting a rod in his mouth) to hear the piano after losing most of his hearing. This was the genesis of bone conduction, which eventually would already be perfected in the 20th century for hearing aids and medical applications.

An interesting overview of the many inventions, applications, and patents using some form of bone vibration and actuators is available in a continuously updated article written by John Mortensen (see Resources), discussing the hearing aid pioneers, Bell Labs and Sonotone, and the patent disputes.

During the 1970s, a Swedish medical team perfected the idea of implanting a vibration plate directly into the mastoid bone (the bone behind the ear) to transmit to the inner ear where the vibrations can be processed by the brain as sound, and use an attachment connector for an external audio processor. This patented process is known as a bone-anchored hearing aid (BAHA), and has been perfected with transcutaneous devices that are fully implanted under the skin. The sound processor is then connected using magnetic attachments, allowing users to regularly upgrade their systems as technology evolves, without the need for any more surgeries, and more importantly, avoiding the dangers of an open wound.

As an example, Cochlear, a leading implantable hearing solutions specialist recently received U.S. Food and Drug Administration (FDA) approval for the latest Cochlear Nucleus 8 Sound Processor, which is not only smaller and lighter but is also the first behind-the-ear cochlear implant sound processor using the next-generation Bluetooth LE Audio technology—which has clear benefits for these applications.

Bone conduction technology has proven to be effective for users with conductive or mixed hearing loss—basically people with a functioning cochlea but problems with the outer or middle



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ear. It's highly beneficial for users with single-sided deafness, where a bone conduction device picks up sounds on the non-functioning side and sends them as vibrations through the bone to the functioning cochlea on the other side.

Due to the extensive use in hearing aid applications pioneered by the Scandinavian leading brands in this field, bone conduction transducers (the vibration actuator) have been constantly perfected in combination with external processors, microphones, and combined with conventional electrodynamic and balanced armature transducers.

Cochlear, a leading company in implantable hearing solutions, recently received FDA approval for the Cochlear Nucleus 8 Sound Processor, which is now the smallest and lightest behind-the-ear cochlear implant sound processor available in the industry, already using the next-generation Bluetooth LE Audio technology.

The RadioEar B-81 high output bone transducer is based on the Balanced Electromagnetic Separation Transducer (BEST) principle and widely used in audiometry applications because of its higher output level at low frequencies with lower distortion.

Evolution in microphones is a key area for hearing aids, since that's how environmental sounds are picked up to be transmitted to the transducer. The processor is also a vital component, cleaning the signal picked up by the microphones, filtering the key frequencies (e.g., for speech intelligibility), and optimizing the transmission through the skull bone, or through the skin, depending on the design type.

These transducers are now relatively common in consumer applications, evolving from the concept that was perfected in Sweden by Bo Håkansson, professor of electrical engineering at the Chalmers University of Technology, in Gothenburg. Håkansson devised the balanced electromagnetic separation transducer (BEST) a new type of bone conduction driver, which is now used by all the leading hearing aid companies such as Oticon Medical (Demant Group), and offered by Danish company Ortofon, as the holder of the BEST license for conventional bone conductors.

The BEST transducers have managed to expand the frequency range transmitted through bone conduction up to 10kHz, combining a new type of electrodynamic transducer that uses a balanced suspension principle to counterbalance the forces between the vibrating parts and reduce distortion. The design also allows for much smaller transducers that are still able to generate vibrations down to 250Hz with higher sensitivity.

As stated, Oticon is the company that took this technology to the next level in the treatment of hearing loss, but in the last 10 years, the technology was embraced for consumer applications. Since a bone conduction system leaves the ear canal open, this is a preferable experience by many users who cannot stand in-ear contact, or simply are more interested in remaining aware of their surrounding sounds, and are willing to accept the trade-off in fidelity.

Also, as the effects of prolonged headphone use at unhealthy volumes is being documented, bone conduction is being promoted as a much healthier alternative, particularly for children.

In these consumer applications, the product design sometimes combines traditional bone conduction transducers with other external transducers that act outside of the ear canal. Those can combine haptic actuators optimized for extended low frequencies (bass) and also conventional dynamic transducers that are beaming to the area just outside the ear—such as the recent Bose Sport Open Earbuds, a truly wireless design optimized for sports, fitness, and wellness.

With the recent FDA approval of the long-expected rule establishing Over-the-Counter (OTC) hearing aids, the audio industry will continue to explore these combinations and open ear designs. Manufacturers such as Sonion (another stalwart in hearing solutions from Denmark) offers both bone conduction sensors and vibration sensors, microphones, and balanced armature receivers. In 2018, Sonion started a partnership with Valencell to expand the possibilities of using biometric sensors in the ear and optimize designs for in-ear and on-ear applications. Bone conduction pickup sensors (an area we will explore in a separate article) will be key in the future of true wireless earbud and hearable designs.

The Consumer Perspective

Bone conduction technology was experimented with in early headphones using vibration actuators. In 1994, a patent for "Bone-conductive stereo headphones," was granted to H. Werner Bottesch. That patent described acoustical transducers "devised for the delivery of stereophonic soundwaves through the mastoid bone structure of the human skull." This basically inspired the concept for using bone conduction for leisure listening and music reproduction, an alternative to conventional dynamic transducers.

Ten years later, the market had become somewhat familiar with the concept of bone conduction headphones. But it was the transition from conventional wired headphones to wireless that generated attention from consumer electronics companies. Products such as the Audio Bone established the headband form-factor for use in fitness activities in 2008, but remained connected by a wire to the sound source. With Bluetooth becoming more reliable five years later, wireless bone-conduction headphones hit the market, creating a category that has grown ever since.

The concept was instantly embraced by consumers who enjoy fitness, running, and cycling, and obviously commuters who appreciate the resulting safety of staying aware of environmental sounds. Some countries (e.g., the United Kingdom) even passed legislation determining that only open-ear bone-conduction headphones can be used in competitive athletics and any sports that take place in public environments (e.g., marathons).

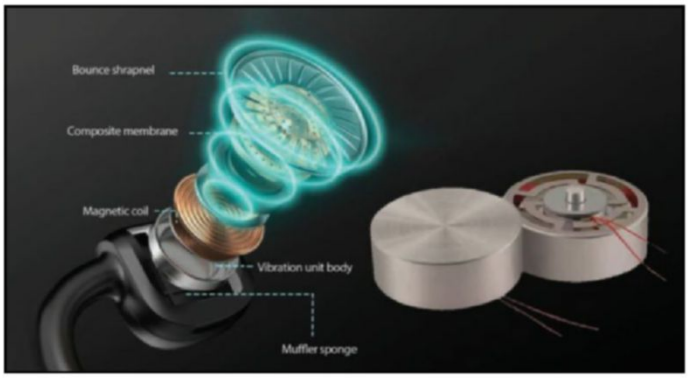
For those activities, a key requirement is the need for the bone conduction designs to remain in place even during intense activity. And the fact that bone conduction needs to wrap around the users' head and create a tight grip is naturally an advantage in those use cases. Of course, other companies have realized that not all users appreciate the force exerted on their heads, and new designs are now more focused on comfort for extended daily use, such as in the office or for home work. The flexible neckbands in comfortable rubber or silicone are now available for heads of all shapes and sizes, allowing users to still appreciate the bone-conduction headphones "hug" around the head, while keeping the transducers firmly in place.

In any case, bone conduction products remain restricted to the headband or neckband form factor precisely because of the need to exert pressure. True wireless bone conduction designs are a challenge, even when using some kind of clamp on the ear. That's why some companies have tried combining bone conduction with other traditional transducers.

Bose's solution to keeping the ear canal open has not taken the bone conduction path. The Bose Sport Open Earbuds features an unoccluded design that leaves the ear canals completely open. They're designed to rest securely on the natural curve of the ear, using acoustic ports to channel sound directly into the ear. Sound waves are sent through the air for a more conventional listening experience by two 16mm drivers—one in each earbud—and a dipole design to minimize the spread of sound, while funneling it down the ear canals.

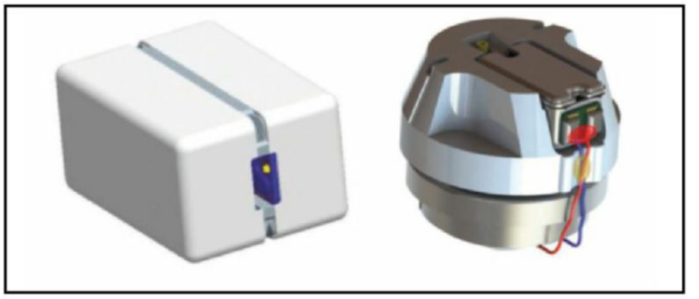
The Sony LinkBuds are another interesting example, introducing an open diaphragm ring design that delivers improved audio transparency at the expense of bass response, yet ideal for augmented reality (AR) content, navigation, and work-from-home applications. The design needs interchangeable ear wings (silicone loop fins) to provide a secure fit.

The search for open-ear awareness is precisely where Auricle, a young startup from Denmark, has focused. The Auricle open-ear design uses two separate bone conduction units connected by an active wire, and focuses on situational awareness. The design



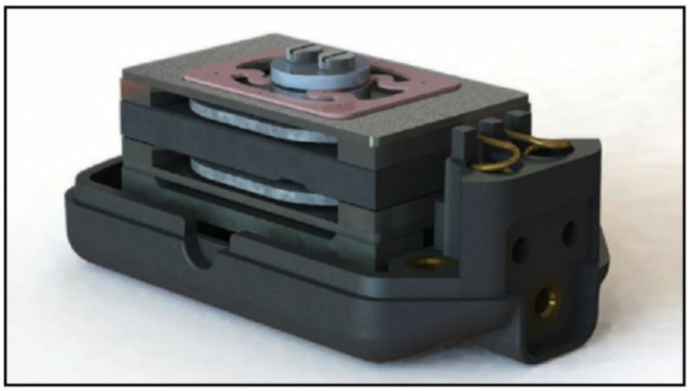
Typical bone conduction drivers combine a coil where the number of windings affects the electrical impedance, a recoil device, or a moving part, two blade springs and magnets, all contained inside a housing, the design of which influences the response due to resonances.

Panasonic had some success with its water- and sweat-resistant RP-BTGS10-k wireless bone conduction headphones, that paved the way for the consumer category.



Sonion, the manufacturer of components and solutions for hearing aids and in-ear headphones, offers voice pick up bone sensors and also offers its miniature magnetic vibration transducers for use in bone vibration applications. The company also offers hybrid solutions with bone conduction.

holds to the tragus, a small and unique cartilaginous structure of the ear, located on the anterior margin of the auditory canal that offers an effective placement for bone conduction without obstructing the ear canal. After extensive testing and research, the company perfected a very light enclosure that is able to remain firmly attached. The tragus grants a backward slope, guarding the opening of the auditory canal, which also proved ideal to hold a clip-on bone conduction transducer. The company is now optimizing the design for a new-generation bone/haptic conduction transducer, able to deliver an extended frequency response down to 50Hz, in order to achieve better fidelity for music reproduction. The design has all the potential to eventually become completely true wireless in the next generation.



Ortofon Microtech offers a range of bone conduction transducers, including specialized versions for use in hearing clinics in combination with headphones for audiometry checks, models for hearing aids, and conventional models for speech or music signals.

EnListen launched specialized headphones for sound listening training based on a popular model produced by AKG, with an added bone conductor inserted in the right ear cuff to send sound signals through the right mastoid bone. A dial could control how much conduction sound goes to the left ear relative to the sound delivered to the right ear. Set at 10, both ears received the same contributed volume.

As Pedro Costa, the company's founder explains, "We have found the placement on the tragus to be the most efficient location relative to the amount of power required to drive the transducer—irrespective of stimulated frequency. In other words, by attaching the auricles to the tragus, we get the most bang for the buck in terms of force (aka how much sound we can transmit via the skin and bone relative to the output wattage/voltage) and frequency range/response 'balance'."

For cyclists and motorcyclists, bone conduction transducers are also now frequently incorporated in all types of helmets, drawing in great measure from the experience in military and aviation applications. In those applications, pressure is naturally exerted by the helmet, even though not always in the right place.

Aside from the ergonomics, bone conduction transducers excel in the human voice frequency range and can easily be optimized for voice reproduction. That helps explain why many of the bone conduction products on the market are popular among consumers who use them for calls and listening to audiobooks and podcasts.

A Haptic Perspective

Bone conduction, with its roots in military communications and the hearing aid industry, has included vibration pickup as an alternative or to augment airborne sound pickup. Haptics create an experience of touch by applying forces, vibrations, or motions to the user. These actuator feedback devices are also needed to create the illusion of substance and force within the virtual world—applications in user and human-machine interfaces, gaming, virtual reality, simulations, and the so-called metaverse. Their functions and locations are truly all over the place with end-users coming into contact with haptics in smartphones/tablets and other touchscreens, wearables in gaming, sport fitness, medical, wellness, and more.

In previous *audioXpress* articles we have explored bass shakers for home theater, music performance, flight simulators, amusement rides, and pilot training, as well as haptic actuators generating reassuring vibrations for wellness or a simple "handshake" in a button to acknowledge our order selection. Yet, by far, the most popular use for haptics are the miniature devices in smartphones that alert us when in our pockets.

Those haptic motors that most of us initially encountered in flip phones used a Eccentric Rotating Mass (ERM) micromotor with an off-center weight on the shaft. The ERM vibration motor has faded, except at the bottom end of the market, as the more sophisticated Linear Resonant Actuators (LRA) vibration motor is now common for higher-grade applications.

Devices such as the TDK Mini PowerHap actuators have been introduced for these more simple user interaction applications. But these traditional haptic devices only deliver simple vibration buzzes due to the narrow frequency range and non-agile response. In contrast, larger panel actuators (under the screen applications) and today's bone conduction devices are transducers that reproduce a vibrational response analogous to the audio signal.

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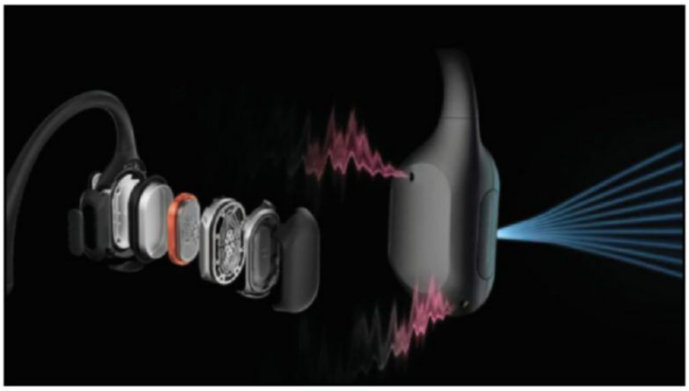
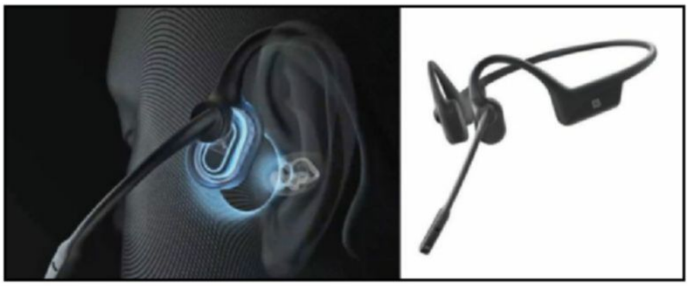
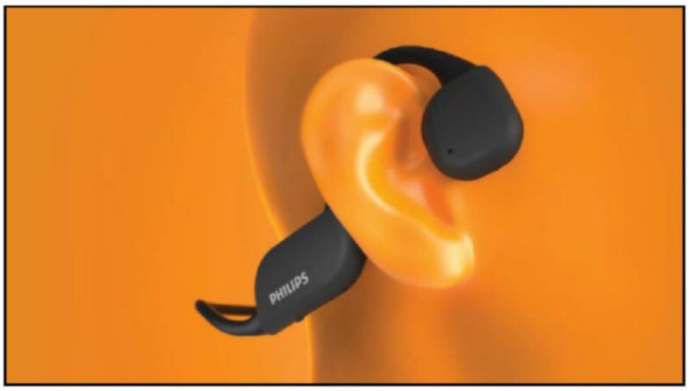


Shokz, the company recognized for its patented bone conduction technology and open-ear headphones, recently launched OpenRun Pro Mini, a smaller version of Shokz's most premium bone conduction sports headphones, the OpenRun Pro. The new design is powered by patented Shokz TurboPitch 9th generation bone conduction technology, which adds two bass enhancers located inside the transducers for improved sound quality and enriched bass, while keeping both ears open to surrounding ambient sounds. In addition, spoken words are softer and more natural-sounding, making OpenRun Pro Mini a great choice for audiobooks and podcasts.



In 2020, Shokz launched the Opencomm headset, in a brilliant move that combined its previous generation model (Aeropex) with a microphone boom. Bone conduction is proven to work very well with voice frequencies (the reason why it's used for military and fighter pilot helmet applications), and since workplace situations and working from home (usually) offers less noisier environments, the combination results in a great alternative compared with convention communication headsets that isolate the user.



Staying aware of the surroundings and being able to enjoy music, take calls, receive instructions, or listen to a reference track while playing on stage. There are many advantages for these open-ear headbands using bone conduction, and larger brands are paying attention.

BeStar Technologies is a recognized haptics and piezo actuator vendor. Aside from a number of haptics devices, BeStar has introduced the ArFi Haptic Analyzer, a complete and unique measurement system to be able to quantify haptics. ArFi is a modular system that provides options to measure vibro-tactile feedback as well as acoustic feedback, separately and in parallel. Its strong advantage is clearly reflected during the design phase of haptics, as it allows developers to quantify their haptic design and tune it to the given application. ArFi displays the measured data and calculates the intensity value that reflects the human perception of the intensity.

In general, when we search for bone conduction products online, we will find numerous "best of" lists that include products that do not use bone conduction. There are several interesting products that may not strictly be classified as bone conduction and have found applications in earphone or headphone drivers. Some examples would be the aforementioned Bose OpenAudio earphones, which use a dipole configuration to project the audio in an open-ear design. Bose OpenAudio technology is also used in the Bose Frames range of sunglasses, allowing users to hear music and even take calls, and they don't need the tight fit that bone conduction would require.

Tectonic Audio has a transducer for the same purpose for off-the-ear headphones (some readers might remember the legendary AKG-K1000 off-the-ear design from a few decades ago). Tectonic's ancestry goes back to the NXT speaker, a flat panel driven by an "exciter." Its present product lines include exciters, and many can also function for haptic applications and the smallest models for earphones.

There have also always been special cases for first responders, military, and other applications that require situational awareness where an unobstructed ear canal is beneficial. Bone conduction devices can be designed to be difficult to recognize by others while keeping the wearer informed. This is ideal for overt operations or just a more discrete look (smart glasses avoid the cyborg look of the early Bluetooth devices).

An example of an innovative specialist developer of bone conduction devices for first responders, military, and the hearing impaired is TEMCO from Japan. A pioneer of bone conduction established in 1985, TEMCO introduced the world's first bone-conduction earphone microphones, communication systems for helmets, bone-conduction sunglasses, and many other form-factors. The company's high-performance devices are small in size, convenient to wear, resistant to shock and sweat, rain-proof, and suitable for soldiers and firemen.

Shenzhen Voxtech is sort of the Chinese version of TEMCO, with its products also focused on military personnel, security forces, and sports enthusiasts. The British have BAE Systems, another company using bone conduction for military helmets and sailing teams.

Most of these bone conduction transducers are voice coil/magnetic motor structures (electro-dynamic) along with a few piezo types. Perhaps very soon microelectromechanical systems (MEMS) capacitive or piezo MEMS devices will be joining the club down the road.

Back a decade ago, AMI, a Bayer Material Science company, attempted to break into the audiophile end of the headphone market with Vivitouch, its electroactive polymer (or EAP in short). A thin sheet that consisted of two electrode layers sandwiching a dielectric elastomer film, and when a voltage is applied, the two attracting electrodes compress the entire sheet.

This actuator provided the physical "oomph" to the conventional headphone speakers, with both a powerful yet subtle contribution. A few years back, AMI was acquired from Bayer by Parker Hamilton, which is focused on industrial applications and Vivitouch is no longer available.

For now, earphone designers can make hybrid combo designs integrating both earphone drivers with bone conduction. Gaming technology has always strived to deepen player engagement and immersion.

Haptics further narrow the gap between gaming and real life and can be used to extend the frequency response perception to the lower frequencies. Mad Kat and Able Planet headphones used Vivitouch, Razer has worked with Woojer and Lofelt, Corsair works with Taction, and Skullcandy offers its own Sensory Haptic Bass solutions. Other gaming headphones have long offered tactile enhancement.

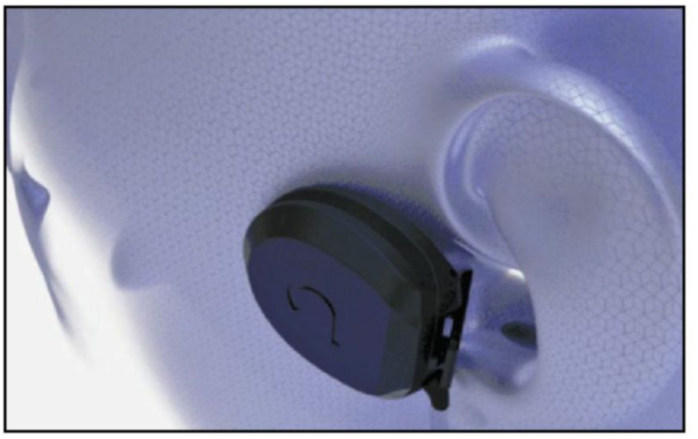
Bone Conduction Brands

Many brands are introducing more refined extended frequency range bone conduction earphone models. From the early days of the Panasonic RP-BTGS10 bone conduction in 2013, all the way to the market leader in the segment that is Shokz (formerly Aftershokz) things have progressed greatly. There are quite a few bone conduction earphones and we can't list them all, but for a sense of the market we will briefly mention some notable big brands and specialist vendors.

Raw bone conduction transducers available in the market for consumer electronics originate mainly with a handful of Asian vendors, where there are also multiple OEM/ODM bone conduction earphone suppliers. We mentioned Danish companies, Ortofon and Sonion, which typically offer solutions oriented for hearing aids but can also provide custom designs for consumer applications. Sonion's bone conducting actuators and sensors can be found in hearing aids and communication, fitness, and lifestyle devices. In the US, Dayton Audio offers the Bone Conduction Exciter (BCE) range that can be used both to drive a surface or in bone conduction earphones and glasses.

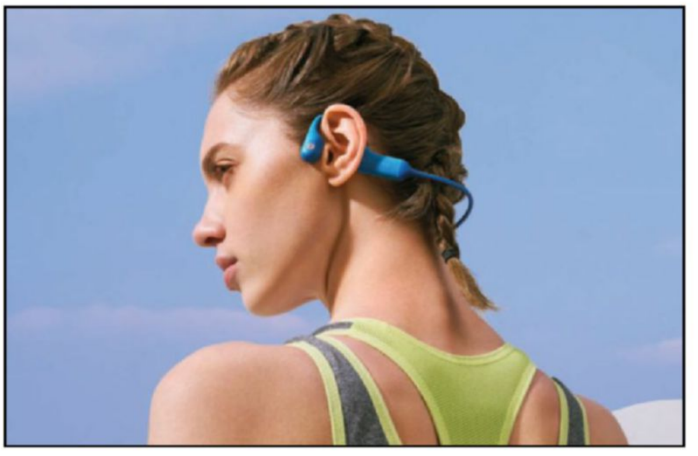
Lofelt from Germany has its patented L5 actuator that is used in gaming applications, such as headphones, with past design wins with Razer and others. The company is now off-the-market, acquired this summer by Meta (Facebook)

Murata Manufacturing from Japan offers miniature vibration devices utilizing the characteristics of piezoelectric ceramic that are low power, have electromagnetic resistance, and support surface-mount device (SMD) mounting, among other performance features not found in conventional electromagnetic vibration devices. Piezo, a division of Midé Technology in the US, is a ceramic actuator specialist. The company offers piezoelectric actuators and motors that can be used in a wide range of haptic applications.



Auricle, a Danish startup, designed an open-ear bone conduction design that holds to the tragus, a small and unique cartilaginous structure of the ear that offers an effective placement and doesn't require the headband from other typical bone conduction designs. The patented design required extensive research and optimization of the transducer type/design and placement.

myFirst, a Singapore-based tech brand, offers bone conduction headphones for children, which provide a safe way to listen to music and watch movies or videos while still protecting hearing health. The vibrations from the bone conduction also allows children's ears to be open to external noise and are designed to sustain rough use.



Haylou PurFree BC01 Bone Conduction Bluetooth sport headphones are a Bluetooth 5.2, IP67 waterproof design with dual-device connectivity for athletic activities.

The Shokz Opencomm UC bone conduction headset features an adjustable DSP noise-canceling boom microphone that also eliminates background noise in loud environments to ensure clear voice transmission in various work settings.

Sound Solutions in Austria—aka Sound Solutions International (SSI) owned consecutively by Philips and Knowles and now part of Foxconn's FIT group, with production facilities in China, R&D in Austria, and offices in the US—is another potential supplier for highly customized applications.

In Asia, Os Electronic is a manufacturer best known for balanced armature drivers but also offers bone conduction transducers for earphones. Reui Men (aka Billion Sound Electronic Technology) from Taiwan offers many off-the-shelf exciters with the smallest applicable to bone conduction earphones. AAC Technologies provides advanced high-performance compact, low power, linear haptic actuators with strong output. Other suppliers with haptic transducers include DLTech Audio, DMEGC Magnetics, and KOTL.

In general, the leading manufacturers in this space have perfected their home bone-conduction drivers in-house in an effort to optimize the frequency response. Panasonic has been producing bone conduction earphones for more than a decade precisely because it develops its own transducers.

Shokz has also developed its own patented transducers. Shokz has a large market share of the bone conduction headphones, regularly updating its range to incorporate the latest Bluetooth wireless technology and features.

The Shokz OpenRun model is highly respected and has received good reviews, which is not so common with bone conduction headphones.

A quick overview of the existing offerings in the bone conduction space reveals an endless number of Shokz emulators from more recognizable brands such as Philips and Orange (yes, the guitar amp company), to a significant number of smaller brands, including Damson, WGP, Vidonn, Tayogo, Sandoo, AKASO, Sentien Audio, Mojawa, and others—all offering very similar designs.

Another emerging bone conduction application is swimming earphones. Shokz has developed the OpenSwim waterproof design for that purpose. Bone conduction is a viable solution for swimming and many existing models are waterproof.

Bone conduction is also becoming common in smart glasses and virtual reality (VR) devices. While effective for this application, the designs require a tight fit between the bone conduction transducers and the wearer, which glass frames usually do not have—or it wouldn't be a comfortable experience.

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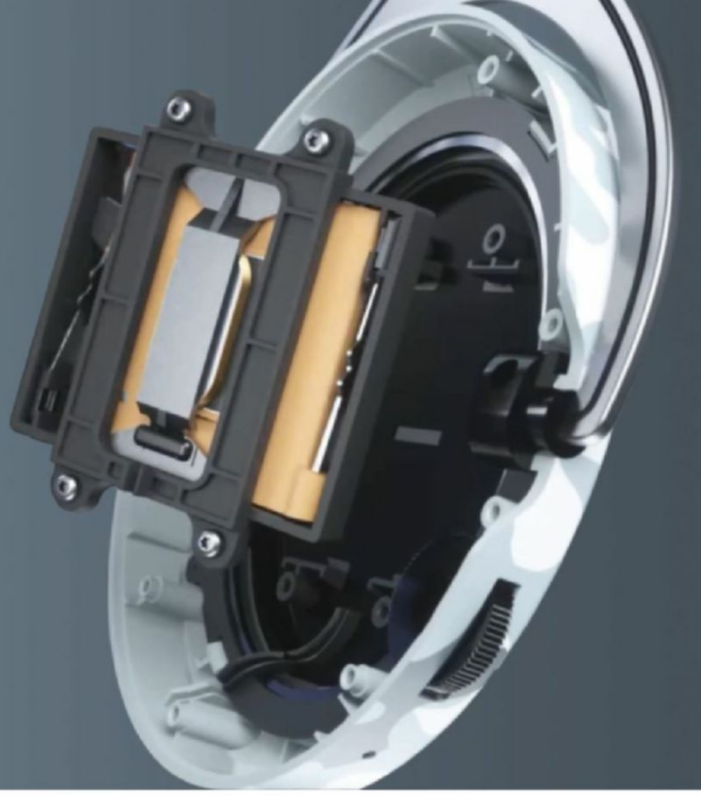
Building a Haptic Dimension

The Science Behind the Taction Transporter



By
Silmon James Biggs and John Steinberg

Haptic motors are finding applications in gaming, music, VR/AR, and therapeutic/wellness. The Transporter is a patented transducer that delivers faster more accurate bass and is designed to work with the main audio driver in headphones. It subtly vibrates against the user's skin to produce an extra dimension that can be described as a high-fidelity extension.

Engineers and audiophiles talk about the spectrum of sound as spanning 20Hz to 20,000Hz. That's only partially correct. Most adults can't hear above 14kHz. More importantly, humans in general don't actually hear 20Hz very well either. But everyone—the deaf and hearing alike—can feel 120Hz and 20Hz, and even lower. In fact, "hearing" deep bass is mostly feeling it.

Bass perception happens along an audio-tactile spectrum. The lowest notes are sensed primarily with the skin, and the highest primarily with the ear, but all of them are naturally perceived as a mix of both senses. That's a big part of the reason why it is so hard to get that "live sound" feeling with headphones—normal headphones can't produce a critical aspect of the live experience.

Does frequency response down to 30Hz, 20Hz, and below matter? The answer is "yes." Rock, hip-hop, and EDM have huge amounts of energy at those levels.

Jazz and orchestral music, plus movies and games do, too. If you've never heard your favorite music on a system that can reproduce ALL the frequencies that are present in the recording, you will probably be shocked to learn how much you have been missing.

So how can you get deep bass? Producing low-frequency acoustic sound at high volumes means moving a lot of air. A good home theater subwoofer can have drivers that are 15" or more in diameter. A PA system for a stadium concert can have hundreds of them.

When those giant speakers crank up, you don't just hear them—you feel them. Your skin vibrates. Your organs vibrate. And (for most people) your brain interprets that vibration as sound, even though it mostly isn't being processed by what you normally think of as hearing.

Perceiving music is like perceiving food. We talk about tasting food, just as we talk about hearing music. But your sense of smell is an essential component of the enjoyment of a good meal. If you have a cold or your sense of smell is damaged, food becomes bland and uninteresting. Just as the input from your nose makes a steak or a fine wine more of a sensory delight, the experience of feeling the bass is what makes going to a concert or dance club so much more powerful than listening to the same music with only your hearing.

Conventional headphones can't deliver that experience—they're like drinking a Cabernet while holding your nostrils closed. (The vast majority of

home and car stereo speakers can't do it either, for that matter.)

Audio drivers in most headphones are (at most) only 1" to 2" in diameter. Some headphones claim to be able to produce sound down to 20Hz. They might even measure acoustically "flat." But your ears are very insensitive to deep bass. For a 15Hz sound to be perceived as equally loud as a 75Hz sound, it has to be 30 times louder—loud enough to quickly cause serious hearing loss, in fact. So headphones that measure "flat" aren't perceived as flat (**Figure 1**).

On the other hand, the human sensory system is extremely sensitive to touch. And it is very sensitive to vibration in exactly the frequency range where hearing lets us down. So a 15Hz signal applied to the skin (depending on the part of the body) only needs velocity 2x higher than one at 75Hz to seem equally strong. The **Figure 2** graph shows touch versus hearing sensitivity and looks a lot like a crossover between a subwoofer and a satellite speaker, doesn't it?

Doing Haptics with Sound

When integrated into an over-ear headphone, the Taction Transporter complements the main audio driver by vibrating the skin to produce deep bass. It does this by subtly moving the cushions of the headphones against our head.

It seems simple, but if done wrong, that movement creates a bunch of problems. Other headphones have tried to do this before, with disappointing results. One problem is that their transducers weren't as good as Transporters. (More about that later in the article.)

How you move the cushions matters too. Most tactile bass transducers to-date have tried to move the headphone cup on the same axis as the audio transducer works, which means moving the headphone cup toward and away from your skull. Moving the earcups this way usually causes serious issues with the frequency response of the acoustic driver, because it dynamically changes the volume of air inside the earcup.

Taction Transporters work differently. The transducers move the headphone cups in the sagittal plane–that is front-to-back along the surface of the skin (**Figure 3**). With this approach, distortion is dramatically reduced. It is also a more efficient way to transmit vibration to your skin. In a properly tuned headphone, Transporters give you epic, subterranean bass without screwing up the acoustic response. But none of that would matter if our transducer wasn't the most powerful, flattest tactile driver ever produced.

Why Flat Frequency Response Matters

How loud should that note be? Musicians and sound engineers try hard to get every sound recorded at just the right level. If your speakers or headphones aren't flat, you hear something

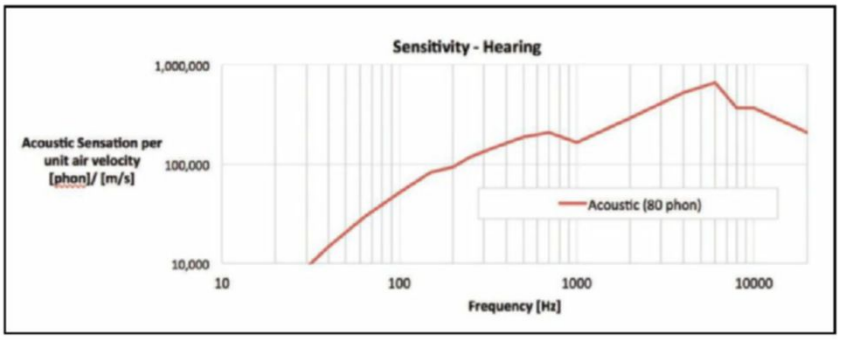


Figure 1: Sensitivity of hearing. At low level our hearing is most sensitive at frequencies between 1kHz to 4kHz approximately, which is 1,000,000 (60dB) more sensitive than at bass frequencies close to 40Hz. Phons represent our hearing experience at different sound levels, described as equal loudness curves. At high sound levels we tend to experience all frequencies at a similar level. The sensitivity of our hearing reduces with increased level and so does the apparent frequency response.

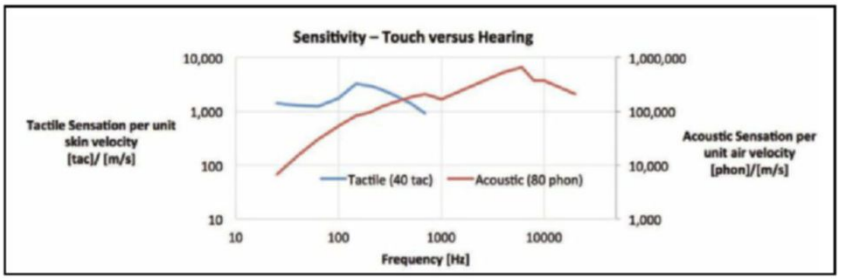


Figure 2: Touch versus Hearing Sensitivity. One tact is the sensation of 1dB of normal skin vibration at 250Hz on the base of the thumb with a contact area of 2.9cm^2, where 0dB is 0.0002 m/s velocity. (Source Image: Curve derived from data in Verillo, IEEE Transactions Man Machine Systems, 1970)

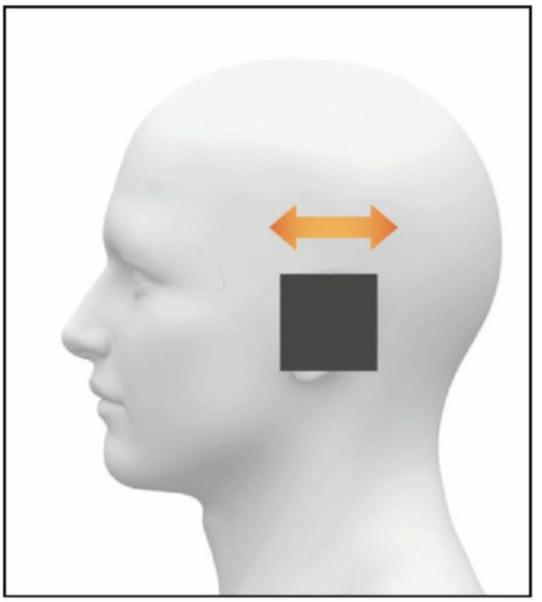


Figure 3: Taction Transporters move the headphone cups front-to-back around the ear, reducing distortion and transmitting vibration more efficiently. This was the subject of Patent No. US10,390,139, describing hardware that can accurately produce a wide range of tactile frequencies at a perceptually constant intensity, moving the skin in multiple axes, and using signal processing to enhance the user's experience of audio spatialization.

different from what they intended. (Note: There is some debate about what the ideal perceptually flat response is for headphones but the disagreements are pretty minor.) Flat frequency response matters for tactile drivers, too.

It's pretty easy to make a low-fidelity tactile bass driver. So some folks have offered headphones that use eccentric rotating mass (ERM) motors (**Photo 1**).

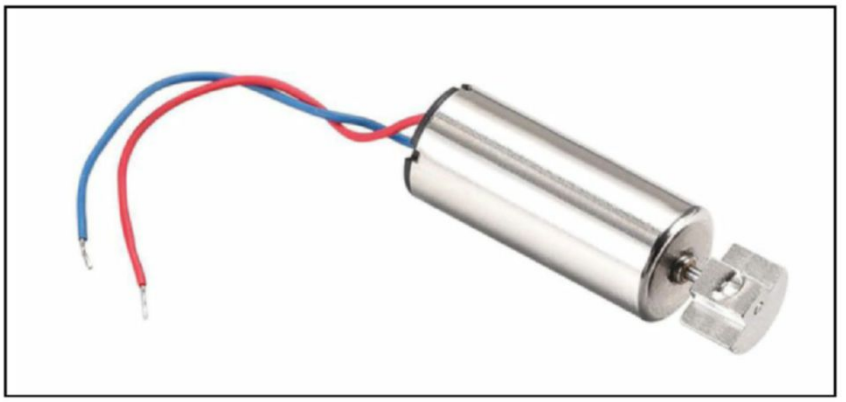


Photo 1: This is an example of an Eccentric Rotating Mass (ERM) motor.

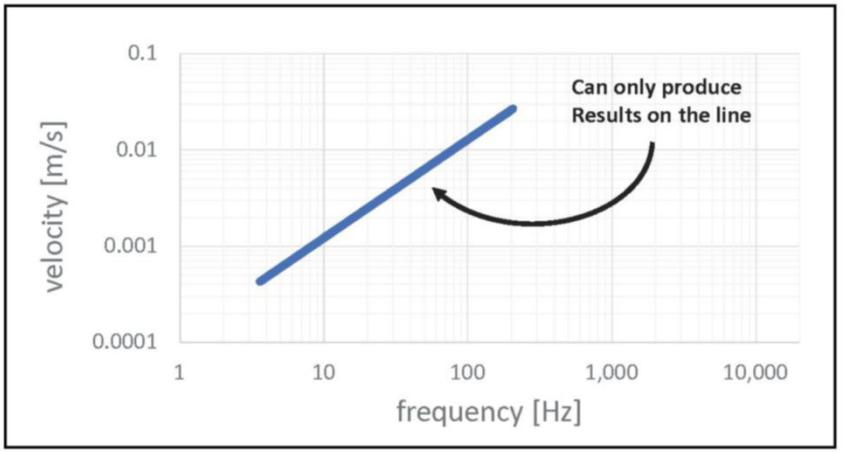


Figure 4: ERM motors can only produce a single volume level for each possible frequency.

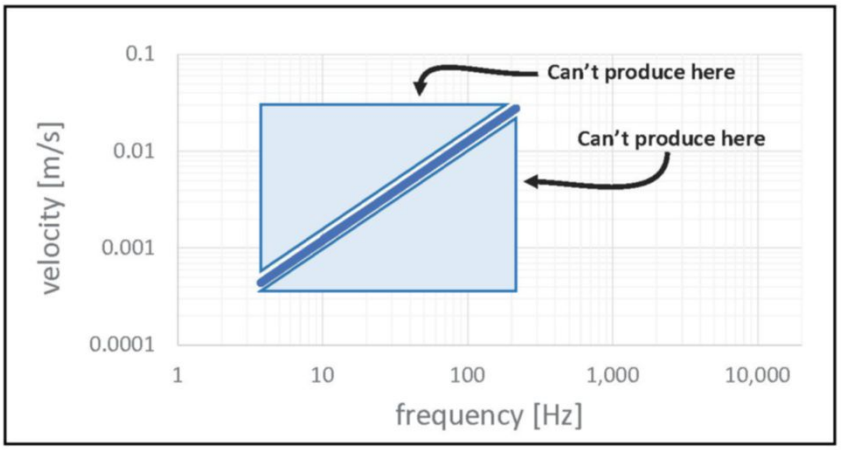


Figure 5: ERM motors cannot play louder or quieter without also changing frequency.

ERMs have been used for years to make cellphones vibrate in silent mode.

A few companies have used them in things like game controllers and headphones (**Figure 4**). While they can produce a range of frequencies, they have a major limitation: There is a fixed relationship between frequency and given power output. If you want it to go louder, it has to rotate at a higher frequency. If you want it to play a lower frequency, it has to get quieter.

ERMs can only produce output along a single line—there is no "area under the curve." So the range of possible outputs from an ERM is quite small, and the effect won't sound much like music for most program material (**Figure 5**).

Another type of device that's been tried in headphones and other devices, including seat-mounted rumblers, is called a Linear Resonant Actuator (LRA). Like ERMs, these can be small and inexpensive (**Photo 2**). But also like ERMs, they are incapable of delivering high-fidelity bass response. The biggest problem with LRAs is in the name: resonance.

Speaker and headphone designers work hard to eliminate (or at least compensate for) resonances so they don't color the sound. LRAs, on the other hand are all about resonance. That makes them very efficient at playing a single note. If an LRA resonates at middle C on a piano, it can play middle C very loud with very little power. But C sharp will be barely audible, as will B natural. And that's all folks—all the other notes on the piano will be inaudible (**Figure 6**).

Some LRAs used in headphones are so resonant that they ring like crazy if you just tap the cup of the headphone with your finger—even when the headphones aren't plugged in.

So how do Transporters compare to LRAs and ERMs? Let's start by comparing them in terms of frequency response (**Figure 7**). Taction Transporters are flat ±4dB from 15Hz to 85Hz. They can produce output at any level from the threshold of perceptibility up to actually blurring your vision at any of those frequencies. Wide frequency response is critical to good tactile performance. But that's only one of several important parameters.

Power That Doesn't Corrupt

As previously discussed, your skin includes sensitive receptors for tactile signals. But there is a threshold below which those signals will go unnoticed. If a headphone manufacturer goes to all the trouble and expense of adding tactile transducers to a pair of headphones, the system had better be capable of producing enough tactile output to be felt.

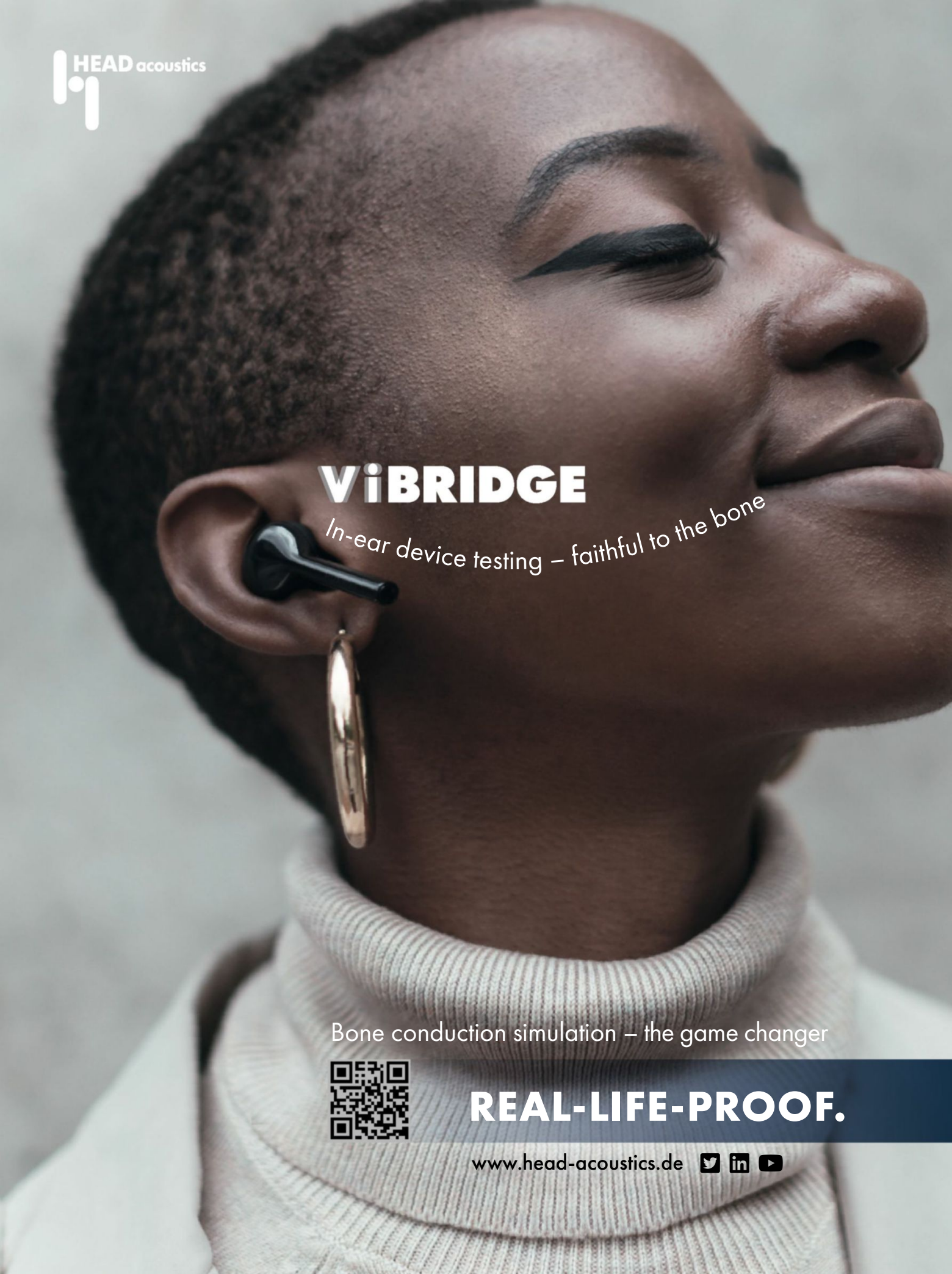

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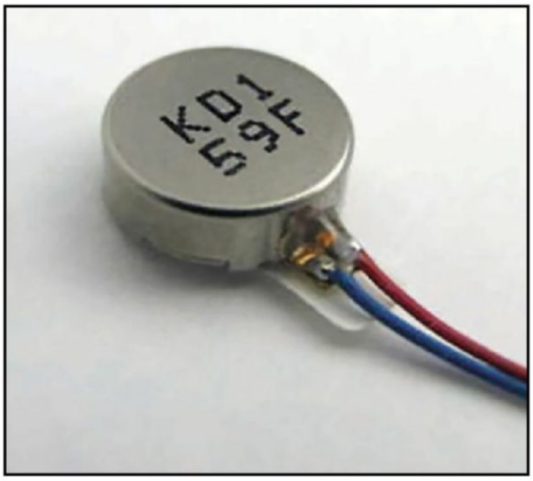
Photo 2: Here is an example of a Linear Resonant Actuator (LRA) driver.



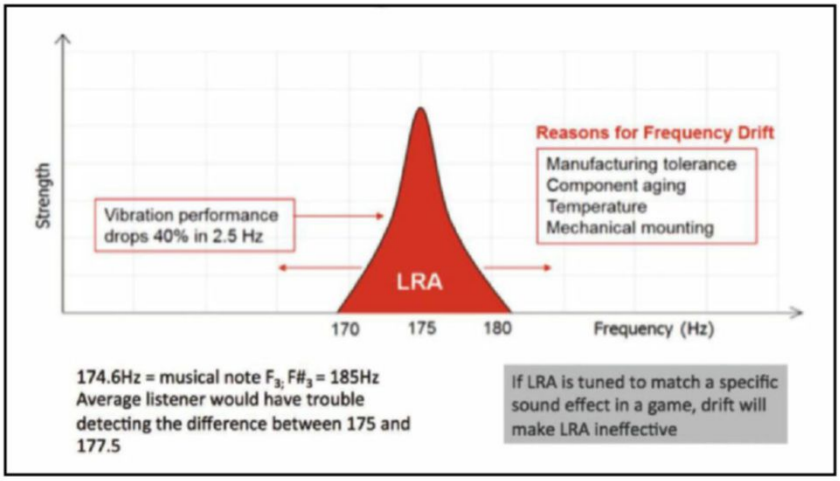

Strength
Vibration performance drops 40% in 2.5 Hz
LRA
Reasons for Frequency Drift
Manufacturing tolerance
Component aging
Temperature
Mechanical mounting
170
175
180
Frequency (Hz)
174.6Hz = musical note F3; F#3 = 185Hz
Average listener would have trouble detecting the difference between 175 and 177.5
If LRA is tuned to match a specific sound effect in a game, drift will make LRA ineffective

Figure 6: The frequency response of a LRA is shown here. These are so resonant that power can drop 40% just 2.5Hz away from the resonant peak.

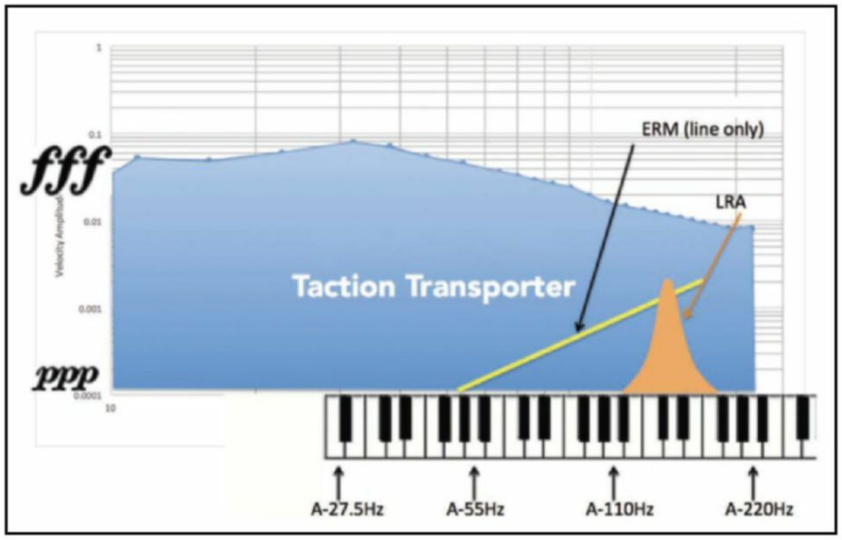


Figure 7: This graph shows a comparison of the low-frequency output among Taction Transporters (blue area), Eccentric Rotating Mass motors (line), and Linear Resonant Actuators.

There are several different ways of measuring power as delivered by a tactile transducer. (Headphone designers, take note: Your head-and-torso simulator will not properly measure tactile output. Tactile transducers don't (intentionally) produce changes in air pressure.) Taction offers dedicated test equipment based on three-axis accelerometers to measure tactile output.

You can measure tactile output as acceleration, as velocity, or as displacement. The measurement that best correlates with user experience, in our opinion, is velocity. A transducer that delivers constant velocity over a given frequency range will be perceived by a user as providing flat frequency response.

So how much velocity is required? The threshold for perception of a tactile signal varies from person to person, and is different for different parts of your body. But a good baseline is 5 mm/second.

The ERMs and LRAs used in other headphones that claim tactile output are severely challenged on this score. The LRA used in one well-known headphone produces a tactile signal above the perceptual threshold only around the resonant frequency. Unfortunately, when it's doing this, acoustically it loudly rings like a bell at about 75Hz.

An ERM could theoretically deliver strong output at the top end of its frequency range, but (a) as previously discussed, the only way to increase the output of an ERM is to increase its frequency, which obviously doesn't deliver flat output; and (b) the ERMs commonly used just aren't that powerful. In a typical headphone, the Transporter can deliver at least 5mm/s of earcup velocity from 13Hz to above 100Hz.

The Need for Speed

Audio drivers with quick rise and fall times make music sound more dynamic, but rise time is especially important for gaming. Saving a few milliseconds of reaction time can mean the difference between killing and being killed. There's research that shows that your body actually processes tactile inputs faster than audio inputs.

But a faster path from skin to brain to action does no good if it takes forever for the signal to become strong enough to reach your skin's perceptual threshold.

Fast rise time is key. Fast fall times matter, too. Three aspects of transducer design are critical to achieving quick rise and fall times: power, mass, and damping.

ERMs have long rise times—on the order of 150 milliseconds or more. LRAs rise faster than ERMs, but only at their resonant frequency. At other

frequencies the rise time is effectively "never." Even so, it takes many cycles for an LRA to generate enough oomph to deliver its strongest output. That's because they tend to have a lot of mass relative to the power of their motors. Transporters are a very different story (**Figure 8**).

The LRA takes about 160 milliseconds to reach 90% of its potential output. That kind of delay is easily noticeable. (Audio-tactile mismatch gets noticeable when its around 40 milliseconds or larger.) That's bad for music. It's even worse for gaming.

By comparison, a headphone, driven by a Taction Transporter can reach 90% of its potential tactile output in just 8 milliseconds. That means the audio and tactile perceptions feel properly synchronized.

So how did we create a tactile transducer with strong output and fast rise times? As Newton figured out, force requires acceleration and mass. Taction brings plenty of both.

To be effective, tactile transducers need to have mass—significantly more mass than other tactile transducers have used. The Transporter's moving mass weighs 18 grams. That's 3 to 10 times more mass than you would find in previous comparable transducers.

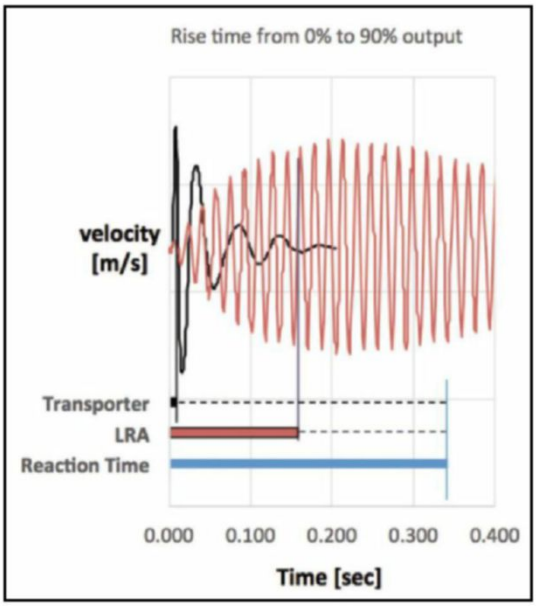


Figure 8: A Taction Transporter reaches 90% of its potential output in just 8 milliseconds. For music, that means the kick of a kick drum feels tight and natural.

Resources

Corsair HS60 Haptic, Stereo Gaming Headset with Haptic Bass, www.corsair.com

dB Enterprises, www.db-ent.com (Sales Contact for Taction: Dave Lindberg (lindberg@db-ent.com)

Taction Technology, www.tactiontechnology.com

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Figure 9: This graph shows a power handling comparison.

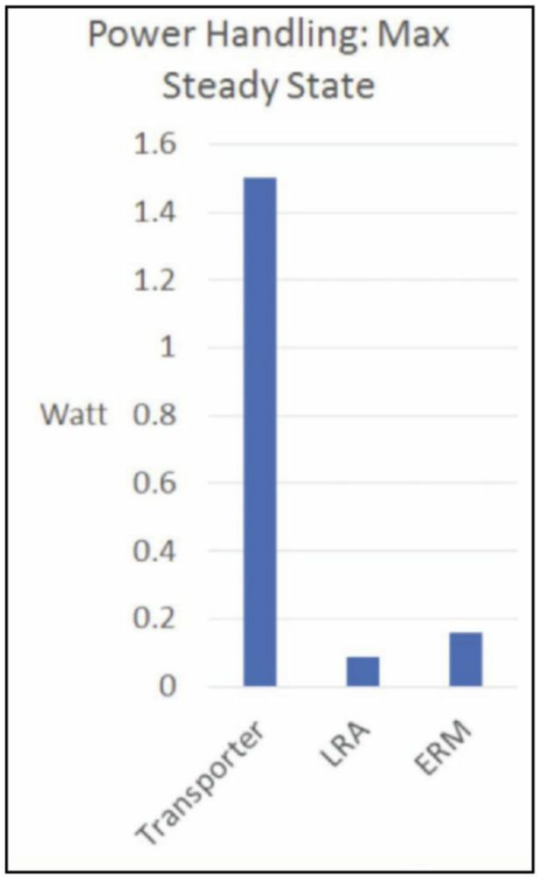
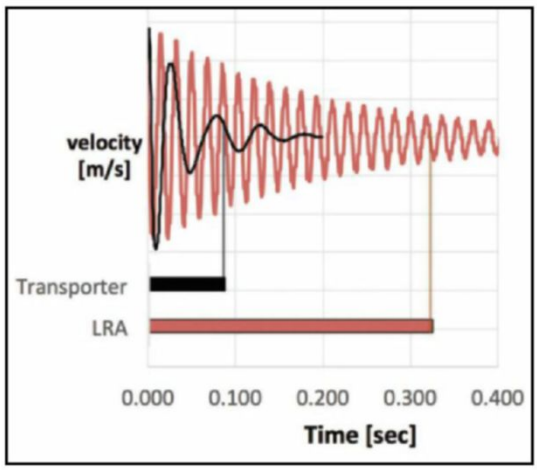


Figure 10: Here's a comparison between the fall time of the Taction Transporter and the LRA used on a competitor's headphone.



To make that mass move quickly, we also need some serious acceleration and that requires a powerful motor. The motor in the Transporter is more powerful than the motor used in any other current tactile headphones we know of and 10 times stronger than the most powerful potentially usable ERM we know of (**Figure 9**).

More mass, plus more acceleration, equals more force. Enough force (at the highest setting) to literally blur your vision. (While most of us might not want that, some folks do, and it provides abundant, low-distortion headroom for the rest of us.) It also makes Transporter transducers viable for a number of other applications outside the context of headphones.

The third critical factor in building a responsive transducer is damping. Damping resists movement. There are many ways to do this, and the properties of damped systems can be very complex.

For purposes of our tactile transducer, you can think of damping as a method for (a) reducing resonances and (b) helping the transducer stop moving, which improves fall times. Transporters use patented damping techniques that give unprecedented control, and excellent rise and fall times (**Figure 10**).

That kind of fall time is critical to reproducing music, and to making sure that sounds like explosions get out of the way of other important cues in gaming. The result, enables the Transporter transducers to offer dramatically improved transient response compared to all previous attempts at using tactile transducers in headphones.

The Most Advanced Transducer

From the extensive research conducted by Taction, we can finally describe the characteristics of the ideal tactile transducer (**Photo 3**):

- Equally strong at all relevant frequencies—from as low as 10Hz to as high as 100Hz or so
- Able to produce any frequency in that range

About the Authors

Silmon James Biggs, Ph.D, is the founder and CTO of Taction Technology, and the inventor of the Taction Transporter haptic transducer. A full-stack tech guru, from mechanical design and digital circuits to software and everything in between, James invested years in finding better ways to interface people with technology. Fascinated by the tactile aspect of sound since he was a kid, it drove him to study human biology at Stanford, and do postdoctoral research at the MIT Touch Lab, including work for DARPA on man-machine interfaces. He also helped develop electroactive polymer transducers for Artificial Muscle, Inc.

John Steinberg is the CEO of Taction Technology, founding CEO of EcoFactor, Inc., a pioneer in the Internet of Things, Big Data, and the Smart Home. John is a Silicon Valley, CA, veteran and a former musician (Berklee College of Music, Oberlin Conservatory). He worked in high-end stereo stores as a teenager and as an adult, treasures his 700+ LPs and vintage audio gear, some of which date back to the 1950s. John has also been issued more than 50 patents.

at any output level—loud, soft, or anywhere in between
- Starts and stops on a dime

Already implemented in commercially available products, such as the Corsair HS60 Haptic Stereo Gaming Headset, Taction now sells off-the-shelf haptic motors or offers a manufacturing license agreement. The published specifications for the Transporter transducer detail the ability to reach the lowest tactile bass note of 12Hz with a velocity of 5mm/s, and reach the highest tactile bass note of 120Hz at that same velocity. Rise time is only 0.008 seconds, and fall time 0.090 seconds. The transducer is able to move the headphones' earcups with a top speed of 50mm/s with minimal impact on headphone acoustics. The intensity is adjustable, maintaining a flat frequency response between 15Hz to 85Hz (±4dB).

Overall Conclusion

We believe the Transporter haptic transducers are able to offer an unmatched deep bass sensation. They are fast, with an instant start and no lag, and they stop instantly, with no ringing.

The simple electronics enable systems using standard audio components. More importantly, the Transporters are quiet, and extend the high-fidelity frequency response all the way to 250Hz.

Photo 3: The Taction Transporter transducer weighs 24g and the device package measures 47.5mm × 40.5mm × 8.0 mm.

GETTING THE MOST OUT OF
VACUUM TUBES
By Robert B. Tomer
Vacuum Tubes—The Important 4/5ths of Tube Circuits
Over 80% of all tube equipment defects result, directly or indirectly, from tube failures. Why do tubes fail? What can be done to prevent them from failing before their time? How can you determine whether a tube is good or bad, or how well and how long it will work in a given circuit? Should tubes be replaced periodically, whether they've failed or not . . . or should they be tested every so often, and replaced if indications show them to be below par? This book supplies the answers to these profound questions…plus many, many more.
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In his refreshing, down-to-earth style, Mr. Tomer shed "Why so many tube types?" "What About Tube Testers?", etc.
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The Benefits of Using Structure-Borne Sound in Headset Signal Processing

By
Hans W. Gierlich

This article details the latest testing strategies for bone conduction technology used in product development for headsets and in-ear devices. It showcases new possibilities for manufacturers to better evaluate device performance, since no appropriate test equipment and methods were commercially available until now.

The use of headsets in communication is increasing, and the demand for improving voice quality in all remote communication continues unabated. While in the past, mostly packaged and corded headsets were used with mobile phones, today's trend is toward advanced wireless Bluetooth headsets in a completely different price and quality range.

Nevertheless, a huge problem is still using such headsets in noisy situations while simultaneously moving the headset microphones farther away from the talker's mouth. This comfort for the user results in a substantial decrease in signal-to-noise ratio (SNR) for the talker's voice signal captured at the headset microphone. The latest technology helping to improve this situation is bone conduction sensors, especially in combination with in-ear headsets. However, testing and optimizing such devices is difficult since no appropriate test equipment is commercially available.

To improve the situation, we complement the HEAD acoustics Head-and-Torso Simulator (HATS) HMS II.3 LN-HEC (with type 4.4 artificial ear according to ITU-T P.57 [1]) with an actuator capable of generating structure-borne sound in the artificial ear. This way, a realistic simulation of the human structure-borne voice signal for in-ear headsets is possible.

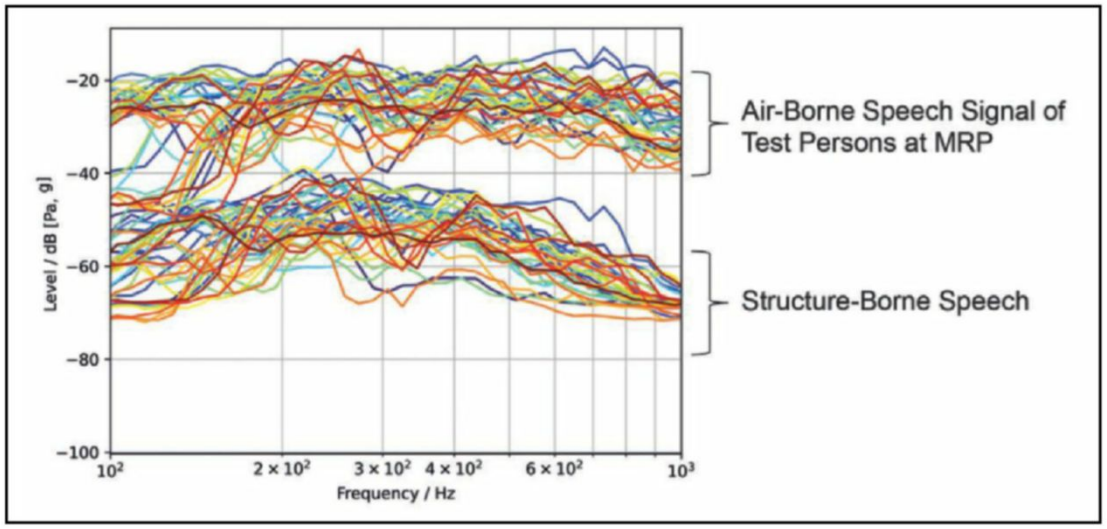

Figure 1: Individual structure-borne signals are compared to the individual air-borne signals at the mouth reference point (MRP) of each test person's mouth.

This article provides background on human structure-borne sound measurement and simulation; on potential benefits of using structure-borne sound as an additional input signal for headset signal processing; and testing strategies and test results.

Human Structure-Borne Sound—Measurement and Simulation

The first step in headset testing with structure-borne sound is understanding the human structure-borne sound transmission of the user's own voice to the user's ear, where the bone conduction sensors of headsets are most likely placed. For this purpose, we implemented a structure-borne sensor in an in-ear headset mockup and performed measurements on a variety of human individuals. The goal is to measure the individual spectra of the air-borne and structure-borne sound. These measurements provide an overview of the spread of individual differences and are the basis for deriving average transfer functions.

Furthermore, we received information on whether there might be significant differences between males and females. As part of the testing, we conducted measurements with 11 female and 24 male subjects. **Figure 1** shows the individual structure-borne signal compared to the air-borne signal at the mouth reference point (MRP). **Figure 2** shows the average of these signals for female and male talkers. The difference shows mainly in the frequency range below 300Hz, where female voices generally do not provide any signal energy due to their higher fundamental frequency. Consequently, we chose the average male spectrum as a target spectrum for structure-borne sound simulation.

The next step is simulating human structure-borne sound. We extended the artificial head HMS II.3 LN-HEC with an actuator, reproducing the average human structure-borne sound in the artificial ear precisely at the location where we conducted the human measurements. **Figure 3** shows the spectrum of the simulated structure-borne sound of the new LN-HEC ear compared to the average human structure-borne sound. Please note that, due to the insufficient SNR of the sensors available today, structure-borne sound simulation currently can only be measured reliably up to 3kHz. As seen in Figure 3, the simulation is correct within a tolerance of ±3dB.

Headset Signal Processing and Potential Improvements with Structure-Borne Sound

The primary purpose of using structure-borne sound in headset signal processing is the improved

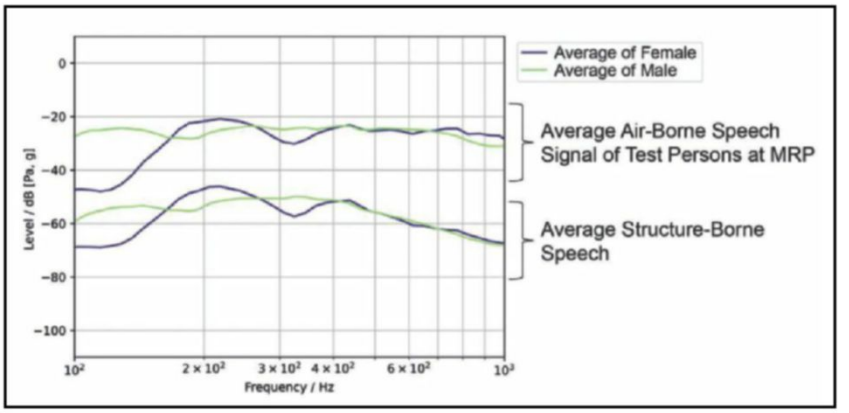
Average of Female
Average of Male
Average Air-Borne Speech Signal of Test Persons at MRP
Average Structure-Borne Speech
Level / dB [Pa, g]
Frequency / Hz

Figure 2: An average structure-borne signal is compared to the average air-borne signal at the MRP.

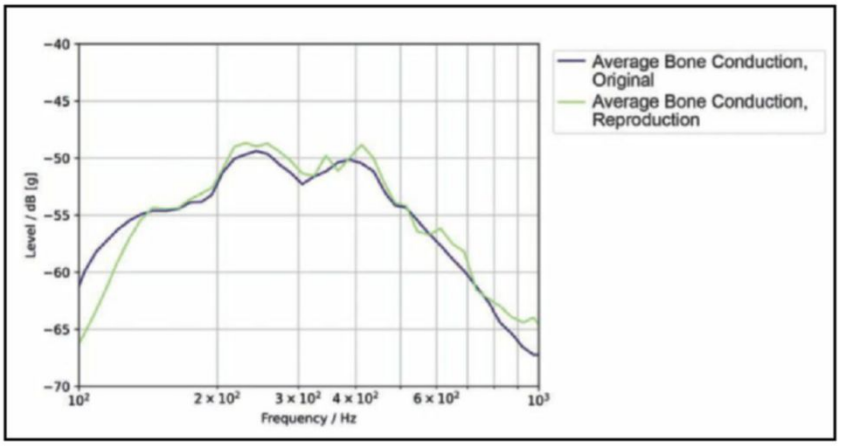

Figure 3: This is a comparison of simulated vs. human average structure-borne sound.

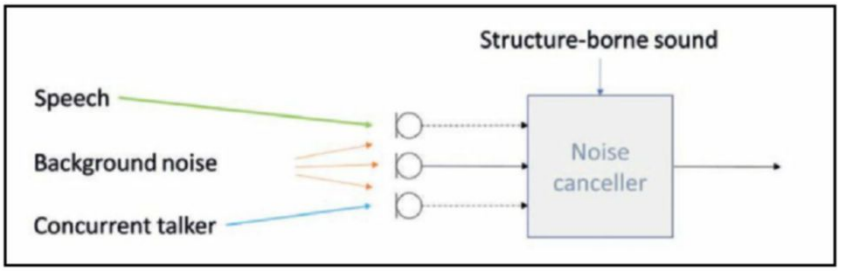

Figure 4: These are signals at a near-end noise canceller.

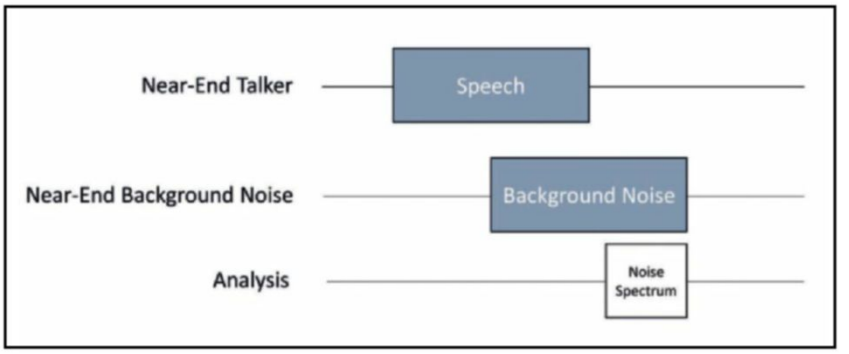

Figure 5: Test sequence A and the corresponding analysis is shown here.

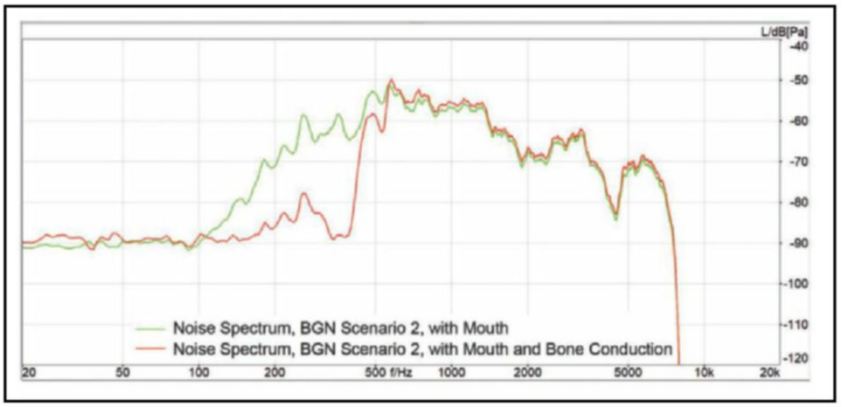

Figure 6: Noise spectra is shown in an uplink with and without bone conduction simulation Device 1, background noise: public noise from ETSI TS 103 224 simulated with 3PASS [2], [4].

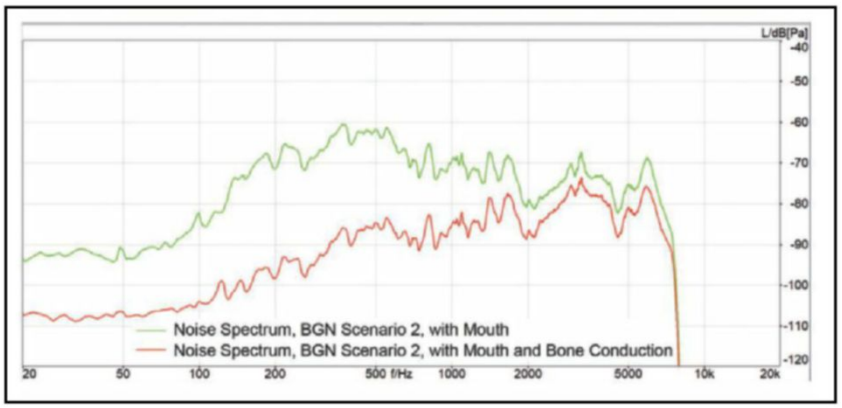

Figure 7: Noise spectra is shown in an uplink with and without bone conduction simulation Device 2, background noise: public noise from ETSI TS 103 224 simulated with 3PASS [2], [4].

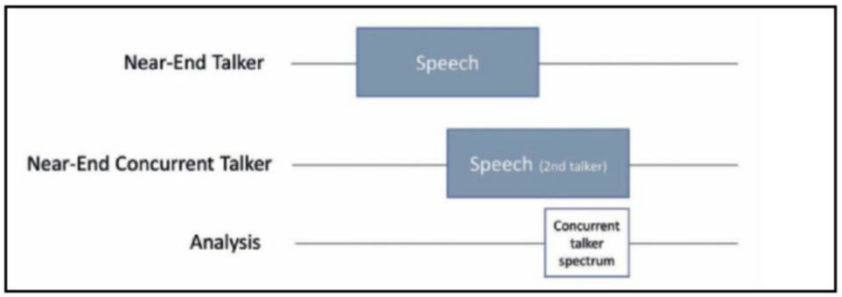

Figure 8: Test sequence A is shown with a concurrent talker and the corresponding analysis.

About the Author

Hans W. Gierlich started his professional career in 1983 at the Institute for Communication Engineering at RWTH, Aachen. In February 1988, he received a Ph.D. in electrical engineering. In 1989, Hans joined HEAD acoustics GmbH in Aachen as vice-president. Since 1999, he has been head of the HEAD acoustics Telecom Division and in 2014, he was appointed to the board of directors. Hans is mainly involved in acoustics, speech signal processing and its perceptual effects, QOS and QOE topics, measurement technology, and speech transmission and audio quality. He is active in various standardization bodies such as ITU-T, 3GPP, GCF, IEEE, TIA, CTIA, DKE, and VDA. He is vice chair of the European Telecommunications Standards Institute (ETSI) Technical Committee for "Speech and Multimedia Transmission Quality" and was chair from 2016–2020. In 2021 he received an ETSI fellowship award.

separation between a near-end talker and near-end background noise, a near-end talker and a near-end impairing talker, and a near-end talker and a far-end talker (double talk detection).

A second benefit of using structure-borne sound could be transmitting the structure-borne signal instead of the near-end microphone signal, at least in the low-frequency domain. The structure-borne signal is almost free of background noise and, therefore, ideally "noise-canceled."

Figure 4 illustrates the signals received at a noise canceller. The separation of near-end speech and impairing signals can significantly improve with a structure-borne signal. What can be expected in terms of enhanced signal processing? The noise canceller is an adaptive filter that estimates the background noise and tries various strategies to reduce background noise by assessing its signal power and trying to minimize the noise level. The better the noise estimation, the better cancellation works.

Separating noise and speech, however, is complex. The noise canceller may diverge, and the speech signal may get degraded. When able to distinguish better between speech and noise based on structure-borne sound, the adaptation control can be much improved. Adaptation can be frozen or adaptation speed reduced, and the noise canceller's divergence may be avoided.

Consequently, a higher-quality speech signal can be provided in conjunction with low background noise. Similar behavior can be expected with concurrent talkers. Assuming only the voice of the talker wearing the headsets should be transmitted, the concurrent talker can be treated as background noise.

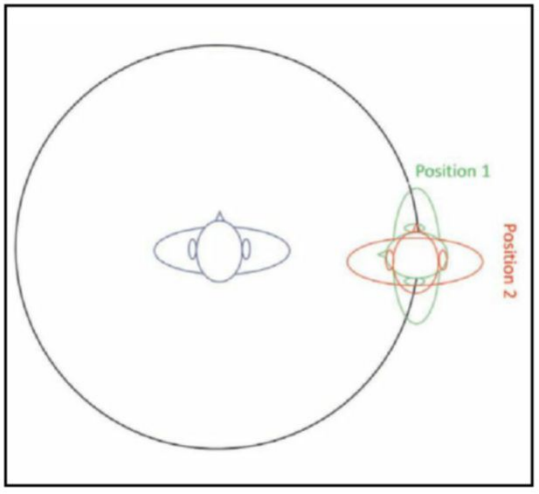

Figure 9: Two positions of a concurrent talker are shown here.

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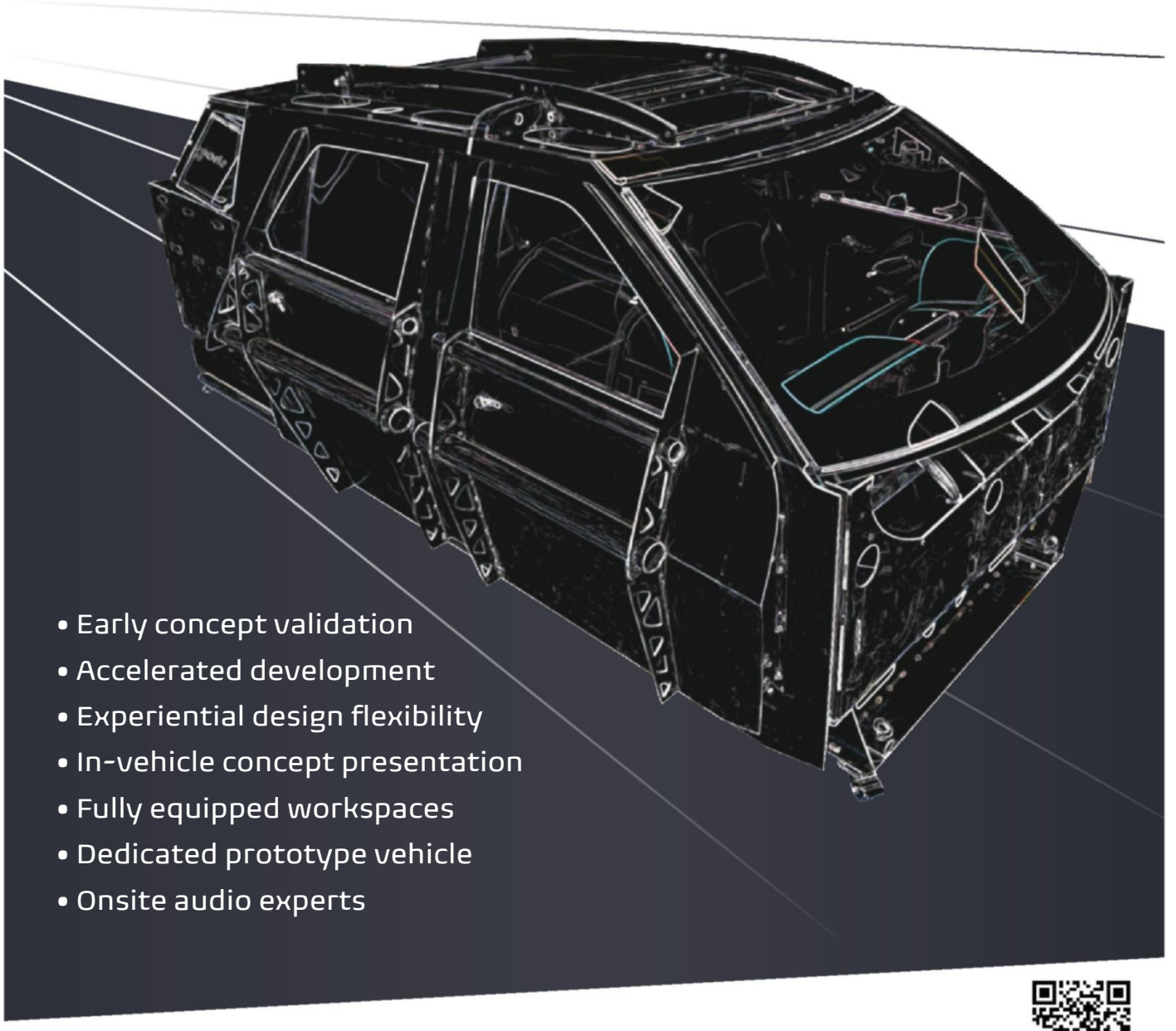

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Measurement Results of Actual In-Ear Headsets

We performed measurements of various headsets based on this fundamental research. The test setup is a typical laboratory-based setup using a realistic sound field simulation with eight loudspeakers and pre-recorded background noises, as described in ETSI TS 103 224 [2]. The HATS is placed in the test room, and the headsets are carefully fitted to the HATS. The speech signals used in all experiments are based on Recommendation ITU-T.P501 [3]. Since measurements in steady-state conditions likely won't show any differences, we created specific test sequences targeting performance differences with bone conduction simulation.

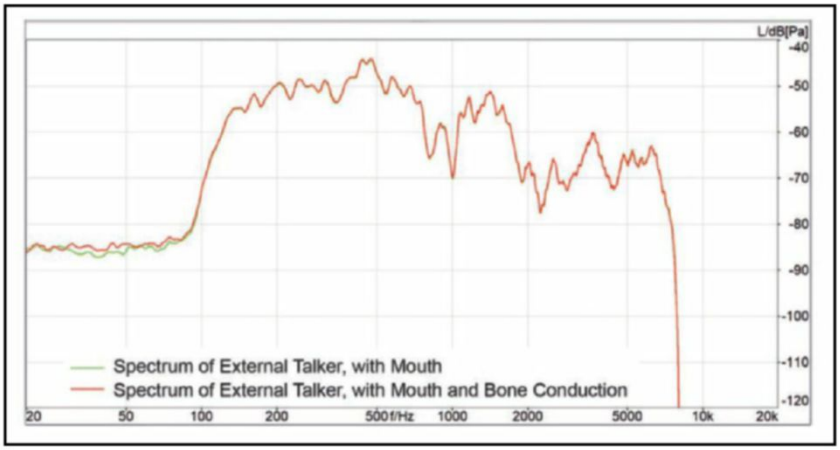
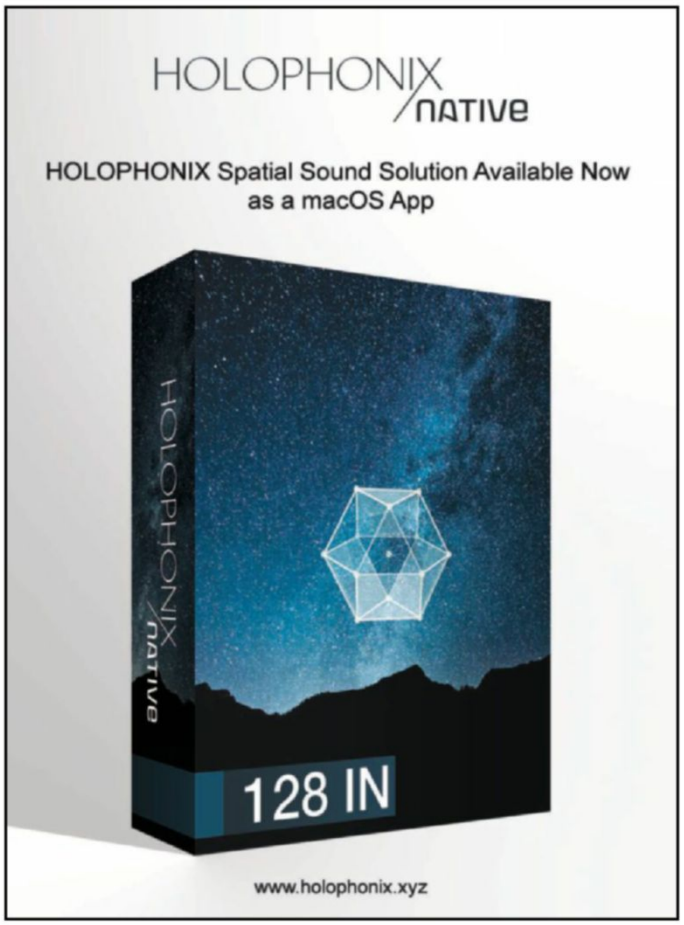


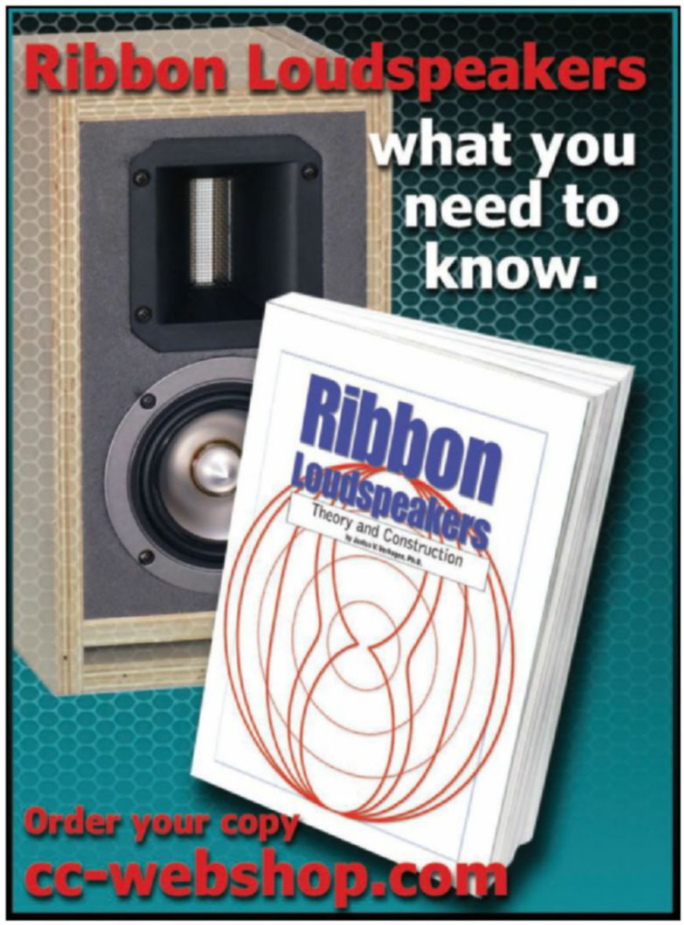
Figure 10: Speech spectra of the concurrent talker is shown in an uplink with and without bone conduction simulation, concurrent talker in Position 1.

Test sequence A, shown in **Figure 5**, focuses on the potential divergence of the noise canceller, which may happen if suddenly a background noise situation occurs. In this situation, speech and noise may not be separatable anymore, and speech might get degraded while the noise is not sufficiently canceled.

Measuring the remaining background noise directly after the speech signal has stopped provides a good idea of the noise cancellation performance. The amount of noise reduction indicates the noise canceller's efficiency in converging reliably even while speech is present. Two example results are given in **Figure 6** and **Figure 7**.

The differences in the remaining noise spectra with and without structure-borne sound are apparent, as well as the difference between the





headset performances. While Device 1 provides an improvement of up to 30dB just in the frequency domain up to 500Hz, Device 2 reduces the background noise by up to 30dB over the complete speech spectrum transmitted.

Experiment B

A similar test sequence can be used to evaluate the performance with a concurrent talker present. For this test, no background noise is used. Instead, a second HMS simulates the concurrent talker. The test sequence for this experiment is shown in **Figure 8**.

You can use various talker positions in this experiment. In this article, we discuss one specifically interesting combination. **Figure 9** shows two concurrent talker positions. In Position 1, the artificial head faces the talker wearing the headsets directly. In Position 2, the concurrent talker is talking in the same direction.

The test results for these two conditions are pretty interesting. While for Device 1, neither a difference between the two positions of the concurrent talker nor a difference between bone conduction simulation nor the lack thereof could be observed (**Figure 10**).

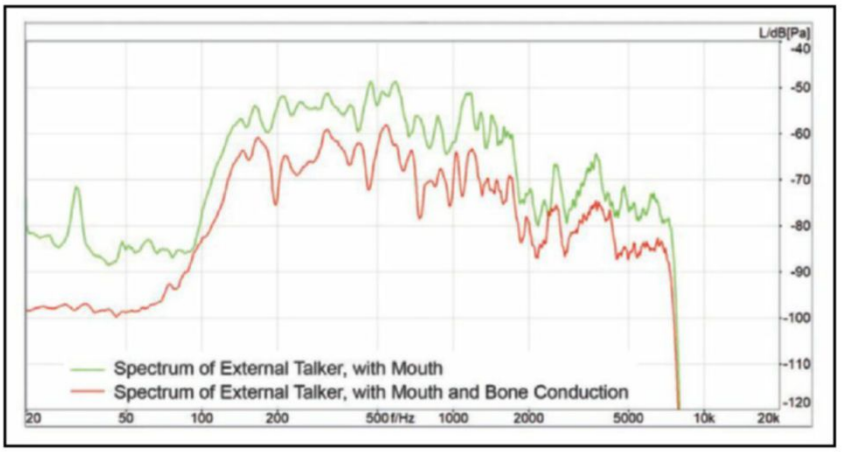


Figure 11: Speech spectra of the concurrent talker is shown in an uplink with and without bone conduction simulation, concurrent talker in Position 2.

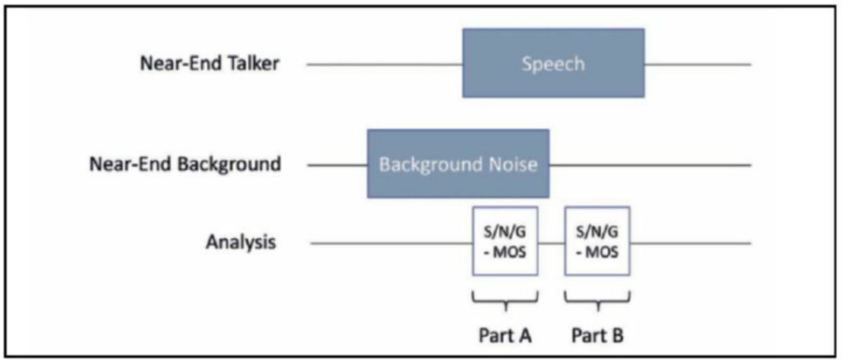


Figure 12: Test sequence B and the corresponding analysis are shown here.

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The result for Device 2 is entirely different. Talkers, facing the wearer of the headset and talking in their direction, are transmitted without any attenuation. When turning 90 degrees, Device 2 differentiates that the second talker may not be talking to the headset wearer anymore and seems to be treated like background noise if bone conduction is present (**Figure 11**). The attenuation is about 10dB.

	Part A		Part B	
Bone Conduction Simulation	Off	On	Off	On
G-MOS	2,4	2,4	2,1	2,5
N-MOS	3,6	3,6	3,7	3,6
S-MOS	2,8	2,9	2,5	2,8

Table 1: S-, N-, and G-MOS results for Device 1 with background noise

	Part A		Part B	
Bone Conduction Simulation	Off	On	Off	On
G-MOS	2,2	2,6	2,1	2,5
N-MOS	3,8	3,8	3,9	3,6
S-MOS	2,6	3,1	2,4	3,0

Table 2: S-, N-, and G-MOS results for Device 2 in the presence of background noise

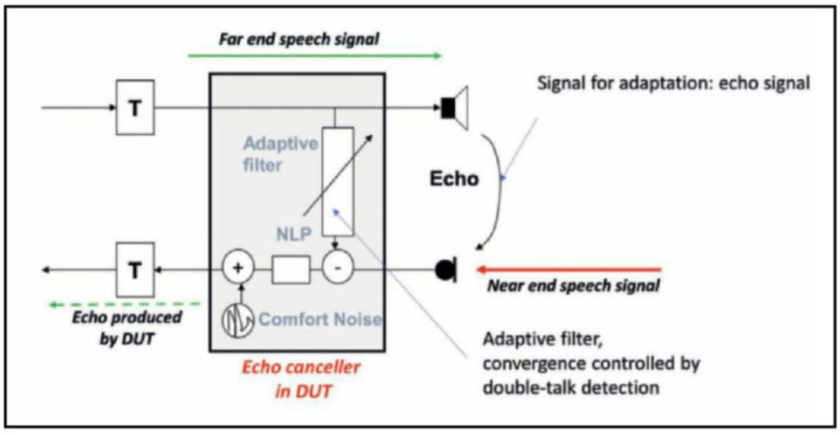

Figure 13: The fundamental functionality of an echo canceller and the associated signals are diagrammed here.

References

[1] Recommendation International Telecommunication Union (ITU) ITU-T P.57 (06/21): Artificial Ears

[2] European Telecommunications Standards Institute (ETSI) ETSI TS 103 224: Speech and multimedia Transmission Quality (STQ); A sound field reproduction method for terminal testing including a background noise database

[3] Recommendation ITU-T P.501: Test signals for use in telephony and other speech-based applications

[4] HEAD acoustics datasheet: 3PASS

[5] HEAD acoustics datasheet: 3QUEST

[6] European Telecommunications Standards Institute (ETSI) ETSI TS 103 281: Speech and multimedia Transmission Quality (STQ); Speech quality in the presence of background noise: Objective test methods for super-wideband and fullband terminals

Experiment C

A third test used the following test sequence shown in **Figure 12**. This test focuses on the noise canceller's potential divergence if speech is presented after background noise. Here we focus on the speech signal (S-MOS) deterioration and a change in noise reduction performance. A perceptually motivated procedure for such evaluations is 3QUEST [5], as per ETSI TS 103 281 [6], which allows for measuring these differences.

The test results (**Table 1** and **Table 2**) for both devices show a clear improvement in speech quality (S-MOS) with bone conduction sensoring. In both devices, the improved speech detection allows for better control of the noise canceller. The speech signal is less deteriorated, leading to higher speech quality in the presence of background noise. This way, devices with bone conduction sensors can be tested and optimized reliably in the lab to tune the noise canceling algorithms best possible.

Experiment D: Double Talk

Another improvement with structure-borne sensing can be expected in double-talk situations. In almost any communication device, echo cancellation prevents the far-end listener from hearing the echo produced at the near-end. In general, the echo canceller (**Figure 13**) is an adaptive filter modeling the echo path and trying to minimize the echo signal by subtracting the inverse echo signal. The echo canceller may diverge if the impairing near-end speech signal is not reliably detected. The adaptation control quality is vital, and the better the echo can be separated from the near-end talker signal, the better the echo cancellation will work in case of double talk. Using the bone conduction signal may improve this separation since it represents solely the near-end speech signal.

Conclusions

Structure-borne sound simulation is crucial when testing modern headsets. Only with the complementing simulation of air-borne and structure-borne sound generated by the human voice can it be possible to evaluate and optimize headsets in different conversational situations comprehensively.

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Acoustical Test Sources

By
Richard Honeycutt

The choice of a sound source for acoustical testing and reporting is decisive and should be determined according to the applications. Let's explore the characteristics and benefits of available sources for acoustical measurements.

Acoustical testing can be classified according to the test setup: Response testing involves a sound-level meter (SLM) and/or a real-time analyzer (RTA) to measure the sound level and/or spectrum produced by a source. Stimulus-response testing incorporates an acoustical source whose sound output is known, and measurements of the resulting sound field are made in order to determine how a room or other enclosure affects the sound. Sound-system performance, reverberation, and echo testing exemplify this type of test.

The second type of testing requires a source with a certain directivity (polar response) and frequency response. The directivity of an acoustical source depends upon its physical characteristics and size.

A source that is small compared to the wavelength of the sound being investigated is called a point source and is omnidirectional. Directivity can be represented in any of several ways.

One way is a simple graph of the beamwidth plotted versus frequency. The directivity graph of an omnidirectional source would be very boring: a horizontal line showing a beamwidth of 360° at all frequencies. Another way is to represent the radiated sound field as a balloon whose shape and coloration represent the sound pressure level (SPL) radiated in each direction.

The third way is a section drawing with a vertical or horizontal cut through the balloon. Such sections often show multiple lines, illustrating the variation in directivity with frequency.

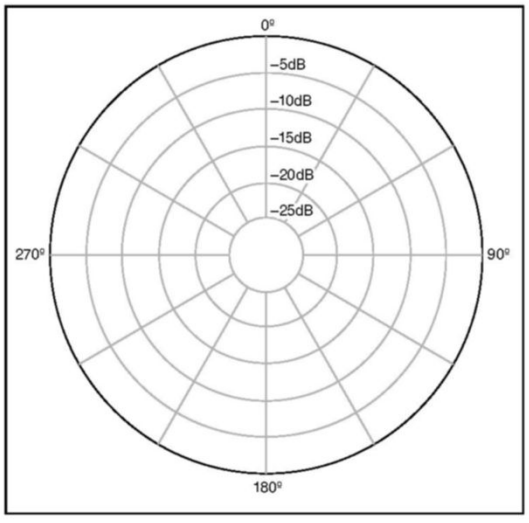

Figure 1: The horizontal and the vertical directivity of an omnidirectional source are perfect circles.

Source Directivity

Figure 1 shows the circular plot characterizing horizontal and vertical directivity of an omnidirectional source. Of course, the balloon plot would be a perfect sphere. **Figures 2–5** show the

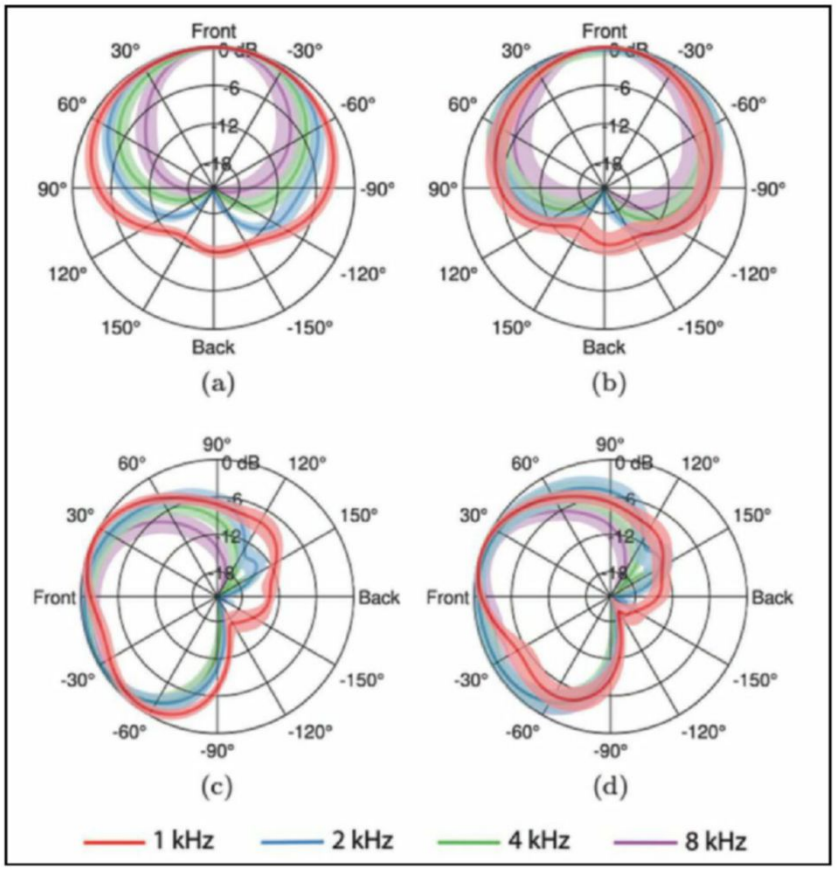

Figure 2: The horizontal (upper) and the vertical (lower) directivity of the human voice vary with frequency; the standard deviation is indicated by the colored shading. (Image Source: Christoph Pörschmann, Johannes M. Arend: "Analyzing the Directivity Patterns of Human Speakers," www.researchgate.net/publication/343344151)

directivities of other common sources. For most of these sources, at lower frequencies, the directivity would be less (the "watermelon" shape of the balloon smaller in diameter); at higher frequencies, greater.

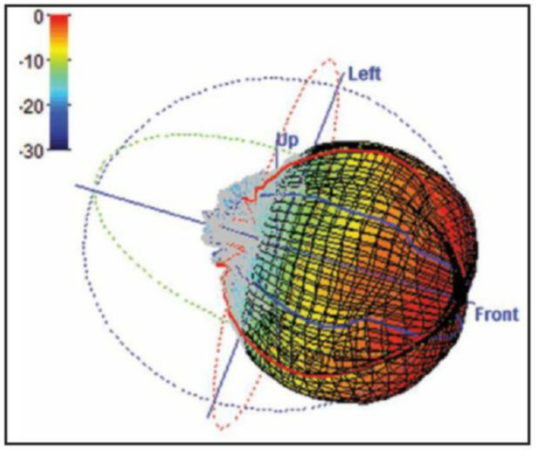

Figure 3: the directivity balloon of an enclosed two-way 8" cone speaker at 3.15kHz shows strong beaming.

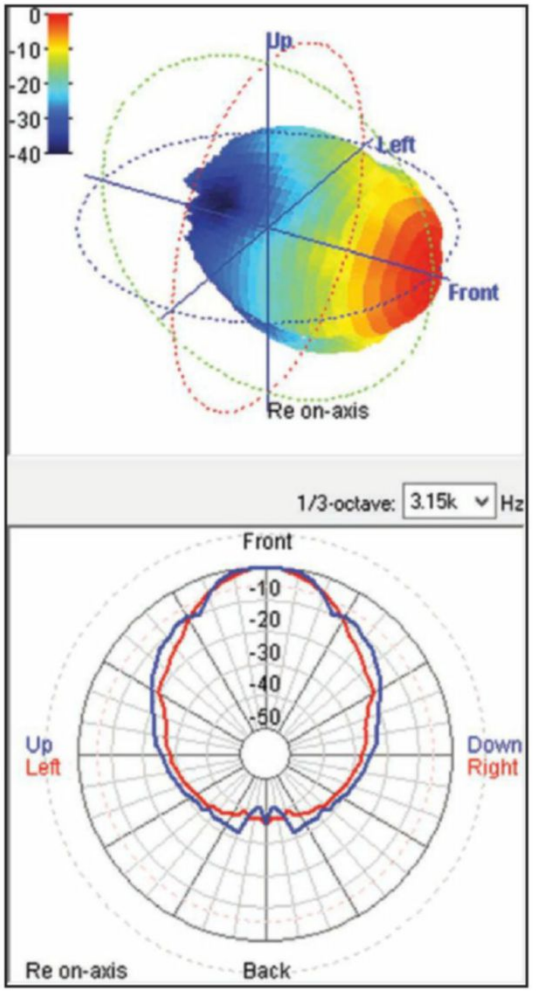

Figure 4: The directivity of a 50-degree horn speaker at 3.15kHz is shown by the balloon (upper) and the vertical and the horizontal balloon sections (below).

For line arrays, the horizontal directivity is typically more-or-less constant at 120° to 150° with some narrowing at the highest frequencies, and the vertical directivity varies from omnidirectional at the lowest frequencies to some essentially constant value (often 10° to 30°) from the lower critical directivity frequency—at which the array height is about ¼ wavelength—to the upper frequency response limit.

For stimulus-response testing of an installed sound system used for reinforcement of speech or singing, a speaker having a reasonably flat frequency response and a polar response similar to that of the human voice can be used. The advantage of feeding a vocal mic using a speaker, rather than simply feeding pink noise to the system is twofold:

- The acoustical characteristics—frequency response and directivity— of the vocal mic, which affect the gain-before-feedback, are included in the test.
- The acoustical environment—stage-monitor leakage and reflections from a lectern table or music stand—are included.

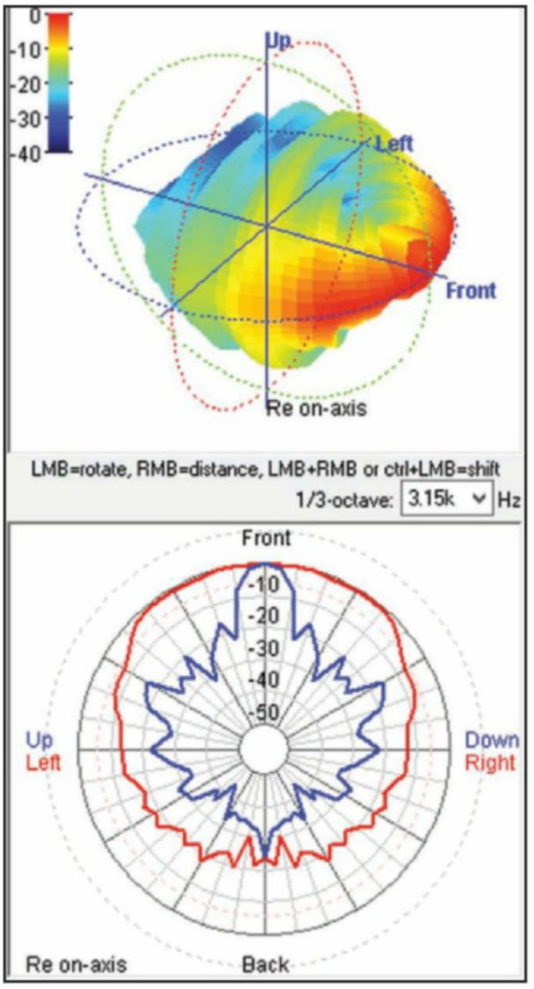

Figure 5: The directivity model of a line array speaker at 3.15kHz is shown by the balloon (upper) and the vertical and the horizontal balloon sections (below).

Photo 1: The B7K mouth simulator can be used to simulate a human-mouth acoustical source.

Photo 2: Another source that simulates a human mouth is the B&K Head-and-Torso Simulator (HATS).

Photo 3: The NTi Talkbox is a high-SPL test source with flat frequency response that simulates the human voice directivity.

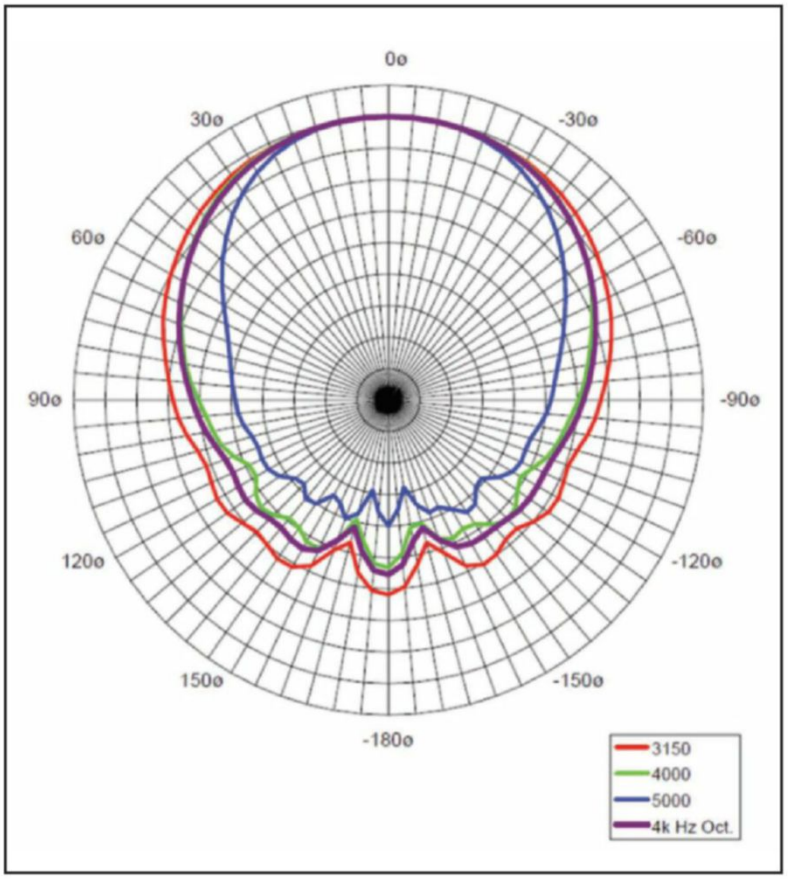

Figure 6: The directivity of the Talkbox in the critical 1kHz to 4kHz octave bands is close to that of the human voice.

Two excellent examples of a test source that can be used in this fashion are the Brüel & Kjaer (B&K) mouth simulator (**Photo 1**) and the Head-and-Torso Simulator (HATS) (**Photo 2**). The HATS provides more accurate directivity compared to a human head. The NTi TalkBox is a less expensive option (**Photo 3**). The polar directivity plots are shown in **Figure 6**.

For measuring reverberation time (RT), an omnidirectional source should be used because the concept of RT as developed by Wallace Sabine assumes perfect diffusion of direct and reflected sound in the room being tested. No real point-source speaker small enough to be omnidirectional in all octaves would be able to produce a high enough SPL to overcome the background noise.

The most common RT measurement is RT30, which is twice the time required for sound to decay by 30dB. For accurate measurements, the background noise must be at least 30dB—preferably 40dB or more—below the test signal, if pink noise is used, as is usually done. (Swept-sine measurements can take advantage of synchronous averaging to cancel out some of the background noise, but not many swept-sine RT analyzers are available.)

This means that test signal levels ranging from 60dBZ to 90dBZ (flat weighting) may be required, depending upon the room noise level. In order to achieve these signal levels, test speakers using multiple drivers are employed. To provide uniform radiation in all directions, 12 identical speakers are often used, mounted in a dodecahedral configuration. The drivers are fed an equalized pink noise signal. The resulting frequency response is not truly flat

in terms of energy per Hertz, but it can be made flat in terms of energy per octave. As sound guru Pat Brown has said, "the object here is to excite the room, not to sound good."

Onmi Sources

Several well-known manufacturers of acoustical test equipment make excellent dodecahedral test speaker ("dodecs"). Examples are the NTi DS3 and the B&K Model 4292L, both of which provide omnidirectional sound at levels somewhat above 120dB. The Italian company Outline has recently introduced the Global Source Radiator (GSR) with a companion radio-remote-controlled white/pink noise generator, 1kW digital amplifier, and other measurement accessories (**Photo 4**).

Commercially available dodecs typically range in diameter from 31cm to 39cm, and weigh from 8kg to 11kg. Most of them are truly dodecahedral in form. The Outline GSR is unique in that it is spherical, reducing diffraction and edge reflections that could cause response irregularities. Good commercial dodecs are expensive, so many audio amateurs and a few audio pros have decided to build their own. A DIY dodec article written by Pat Brown is available online (see Resources).

The Importance of Omnidirectionality

For RT measurements in a room with perfectly uniform dispersion of sound, using an omnidirectional source would not be very important. Such a room would have equally absorptive finishes on all surfaces, and no shapes that could cause focusing or block sound from certain areas. Most real rooms have one or more characteristics that prevent the dispersion from being uniform:

- Ceilings that are more absorptive than the other room boundaries (e.g., acoustical tile ceilings in classrooms and conference rooms

Photo 4: The Outline HSR dodec and its companion amplifier provide omnidirectional radiation at over 130dBSPL.

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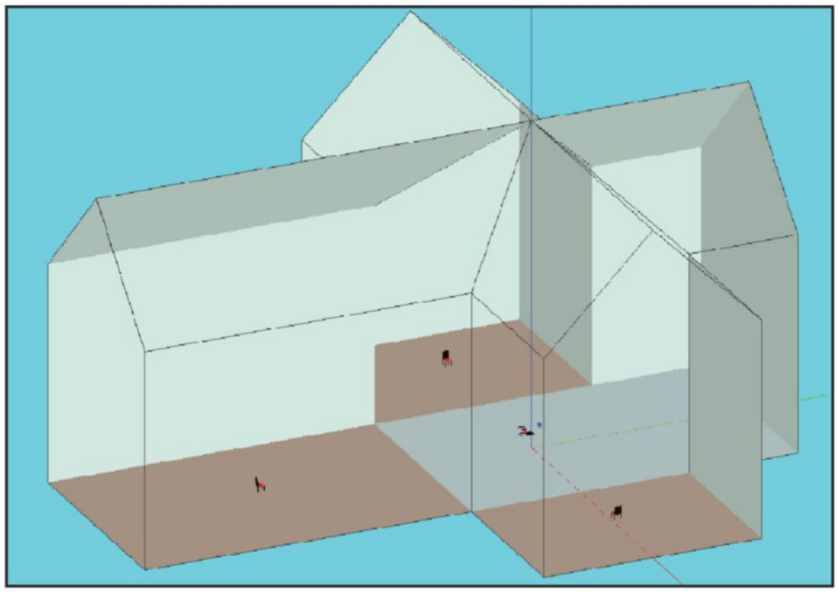


Photo 5: This computer model of a cruciform church sanctuary has several features that call for the use of an omnidirectional source for measuring RT.

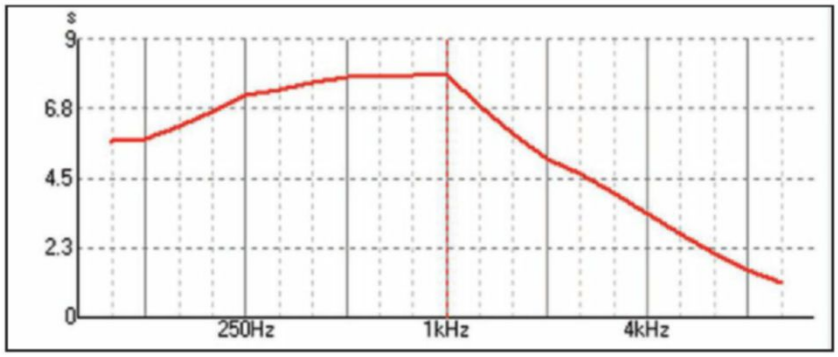


Figure 7: The RT versus frequency (octave bands) measured using an omni source is shown above. The 500Hz RT is 7.72 seconds.

Resources

P. Brown, "Build Your Own Dodecahedron Loudspeaker," SynAudCon, March 2010, www.prosoundtraining.com/2010/03/17/build-your-own-dodecahedron-loudspeaker

"Technical Data TalkBox," NTi Audio, www.nti-audio.com/Portals/0/data/en/TalkBox-Specifications.pdf

Sources

High-Frequency Head and Torso Simulator | Type 5128
Brüel & Kjær | www.bksv.com

Mouth Simulator | B&K 4227
Brüel & Kjær | www.bksv.com

NTi TalkBox
NTi Audio | www.nti-audio.com

- Most of the total absorption provided by an audience located near the floor (auditoria)
- Highly asymmetrical shapes (long, narrow rectangular rooms or rooms whose plan-view form is irregular)
- At least some glass walls (many modern conference rooms)
- Opposing acoustically reflective walls of glass, gypsum, or masonry
- Wall or ceiling areas that focus sound (concave wall or ceiling sections)

In such rooms, an omnidirectional source will provide more reliable RT measurements in some or all octave bands than will a directional source.

As an example, consider the cruciform church sanctuary model shown in **Photo 5**. The tan-colored floor areas represent pews, two-thirds of which are occupied by people, meaning that most of the acoustical absorption is near the floor. The longitudinal and transverse arms of the sanctuary are long and narrow. The peaked ceiling reflects sound toward the center.

Figure 7 shows the RT versus frequency using an omnidirectional source. At 500Hz, the RT at various points in the room ranges from a minimum of 7.78 seconds to 7.87 seconds. **Figure 8** shows the RT of the cruciform sanctuary measured using a full-range horn as a test speaker. This speaker's significant directivity begins at about 200Hz, reaching 50° × 50°at 1kHz and above.

In the computer model, the speaker is aimed at the rear center seating section. At 500Hz, the RT at various points in the room ranges from a minimum of 7.84 seconds to 8.12 seconds. In this room, using a directional speaker rather than an omni as the test source results in negligible difference in the measured RT.

Modifying the room model by adding 2-1/8" cloth-wrapped fiberglass panels on the rear walls and the end walls of the "crossbar" introduces more inhomogeneity in the boundary absorption of the room, and changes the RT range to 4.29-5.03 seconds for the spherical source, and 4.17-5.71 seconds for the horn source, a difference of about 9% in average RT. So the difference in RT measurement results is greater when the room is less diffuse. In such a room, measurement errors resulting from the use of a directional source can be significant.

Normally, the sound source for RT measurements is placed at the location from which the sound normally emerges when the room is in use. For example, in an auditorium, the source might be placed about head height on the stage. In a simple

rectangular room, placement of the source at a corner can excite up to eight times as many room modes as center placement.

In any case, the distance between the microphone and the sound source must be significantly larger than the critical distance of the room—the distance at which direct and reverberant sound pressures are equal—given approximately by: $D_C \approx 0.1(V/RT)^{1/2}$ where V = room volume in m^3.

Overall Impressions

So you need to pay close attention to the performance of an acoustic test source any time you make a stimulus-response measurement. Because having a flat frequency response matters. If you are measuring RT for design purposes, the people, clothing, and number of occupants introduce so much variability in a venue's actual use that precise measurements of the empty venue are often not important enough that a omnidirectional source must be used. However, if you are making measurements to verify compliance with a standard, it is best to use an omnidirectional source.

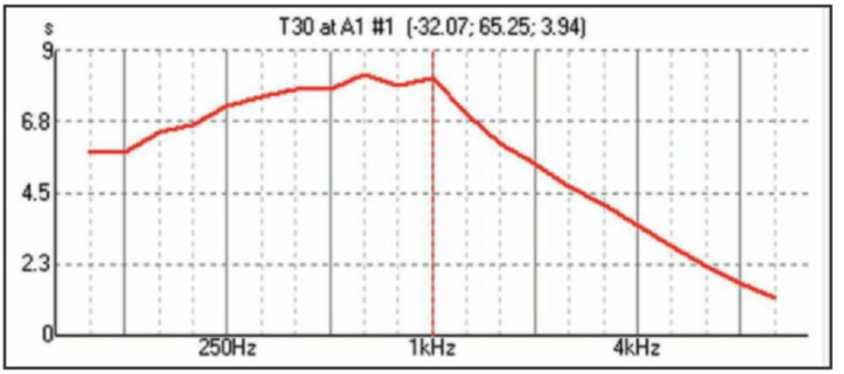
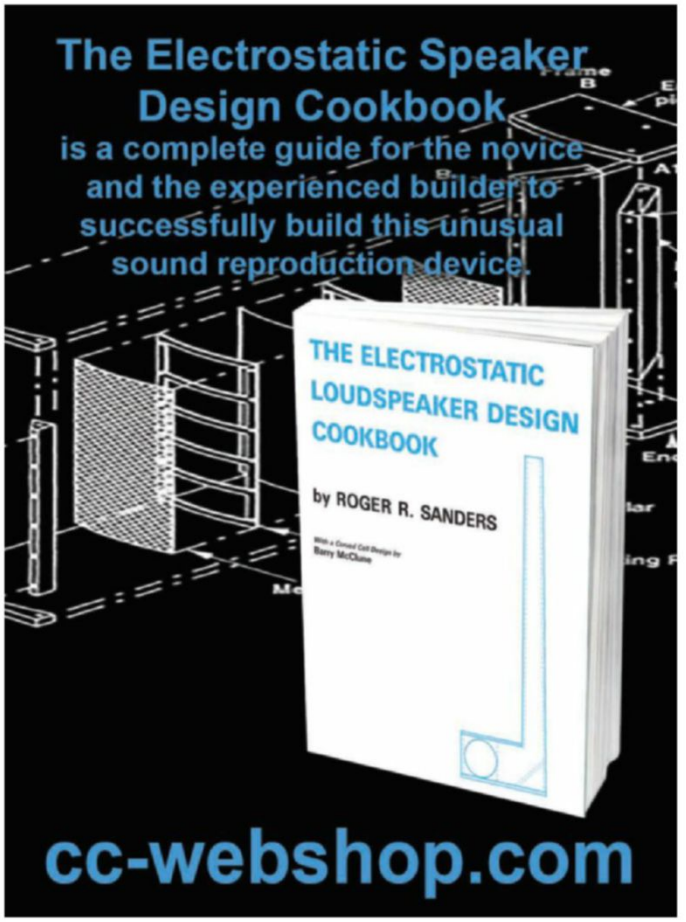


Figure 8: The RT versus frequency (octave bands) measured using a 50°×50° horn source is shown above. The 500Hz RT is 7.72 seconds.

About the Author

Dr. Richard Honeycutt fell in love with acoustics after his father brought home a copy of Leo Beranek's landmark text on the subject when Richard was in the ninth grade. Richard is a member of the North Carolina chapter of the Acoustical Society of America. Richard has his own business involving musical instruments and sound systems. He has been an active acoustics consultant since he received his PhD in electroacoustics from the Union Institute in 2004. Richard's work includes architectural acoustics, sound system design, and community noise analysis.


The Electrostatic Speaker Design Cookbook
is a complete guide for the novice and the experienced builder to successfully build this unusual sound reproduction device.
THE ELECTROSTATIC LOUDSPEAKER DESIGN COOKBOOK
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OTC Hearing Aids

Is Innovation Leaving the Rule Behind?

By
Andrew Bellavia

A wave of new developments promises to solve the most pressing need for people with milder hearing loss, allowing mass-market hearables to supplant over-the-counter hearing aids with less stigma.

To much fanfare in the US, the Federal Drug Administration's (FDA) over-the-counter (OTC) hearing aid rule took effect in October 2022. Devices are now available both in stores and online. This is the culmination of a process that properly began five years earlier with the passage of the law requiring the FDA to create such a category for self-perceived mild to moderate hearing loss. Five years being an eternity in the wearables world, in a March 2022 article [1], I suggested that the assumptions used to write the rule were becoming obsolete even as the process was concluding. A fresh look at developments illustrates how quickly this may come about. CES will be a showcase for the initial suite of OTC hearing aids, while at the same time providing clues to the coming post-OTC revolution.

Ground Shifting Under Our Feet

What we typically think of as a "hearing aid" is a device meant to compensate for damage to, or deterioration of, the cells of the inner ear that convert sound into impulses sent to the brain. Damage is usually selective, causing the ear to be more or less sensitive to sound at different frequencies. To compensate, a hearing aid can be programmed by a hearing care professional to provide varying levels of amplification at different frequencies according to an individual's hearing loss profile. A modern hearing aid is a sophisticated device, providing more control than just the amount of amplification, but selective amplification remains the core feature (**Figure 1**).

The new FDA rule defines an OTC hearing aid in the same way, specifically as a device that "must allow the user to cause frequency-dependent changes based on the user's preference" [3]. An OTC hearing aid also provides control over the amplification profile but with settings determined by the end user rather than a hearing care professional, either through a hearing self-test or choice of profiles.

An unstated premise is that a device's core functionality is determined by design and therefore readily identified as a hearing aid out of the box. The FDA barely recognized the possibility that additional software might change the nature of a device, considering mainly that a traditional hearing aid might be given self-fitting capability later on by unlocking or downloading a program. Therefore, two basic principles undergird the OTC hearing aid rule:

- Selective amplification is needed to ameliorate mild to moderate hearing loss.
- Such amplification must be provided for at the time of manufacture, even if a self-fitting component is activated by software afterward.

Both of these will prove untrue, bypassing the OTC rule in different ways and opening the door to increased innovation in hearing devices.

Amplification and the Cocktail Party Problem

People with more severe hearing loss require selective amplification to understand other people even in quiet settings. Others whose hearing loss is milder, or even nonexistent according to a standard hearing test, often hear fine in quiet settings but have difficulty in loud restaurants or pubs. This "cocktail party problem" is one of the most difficult because the noise is created by other voices. The ability to discriminate between the voice one wants to hear and all the others is one of the most sophisticated functions of the entire auditory system from ear to brain. It doesn't take much to impair this ability, which is why a significant number of people without measurable hearing loss report difficulty hearing in noise (**Figure 2**) [4],[5].

That leaves open the question if amplification is even necessary for people who are challenged only in loud situations. All that

is lacking is an effective way to separate nearby speech from the din. The company that solves this problem will bypass hearing aid regulations entirely, opening up the possibility of offering "restaurant mode" in mass-market true wireless stereo (TWS) earphones. Even Apple has moved in this direction, adding by software update what it calls "conversation boost" (directional microphones) and ambient noise reduction to AirPods Pro. With up to 7dB of signal-to-noise ratio (SNR) improvement in hearing [6], these modes offer meaningful benefit in a popular consumer device (**Figure 3**). And that is just the beginning.

Traditional hearing aids improve speech in noise performance in similar ways to AirPods Pro, using a combination of directional mics and noise filtering. Referring to Figure 1, hearing aids are more sophisticated in that they perform amplification, compression, and noise reduction on per-channel basis to squeeze out higher SNRs than is possible operating on the entire spectrum as a whole. In addition, sound scene analysis dynamically controls the filter bank as the auditory situation changes. Because the noise profile in a restaurant is mainly the babble of other voices, there are limits to the efficacy of acoustic filtering, even under dynamic control.

A Solution from the Broadcast World

Those in the broadcast world are familiar with machine learning-based noise reduction through programs, such as Krisp. These work on a completely different principle. Rather than performing acoustic filtering, machine learning systems are trained to separate speech from noise and pass on only the speech. Often a deep neural network is used. This is a method of analyzing an input similar to the way the human brain works. Just as a person can learn to recognize the difference between a cat and a dog from life experiences, a deep neural network can learn the difference between speech and noise by being trained with speech recordings in different scenarios. The result is an inline system for separating out the speech and sending it through without the noise (**Figure 4**). Krisp wasn't even launched until 2018 and still requires a PC's processor and resources to create the deep neural network in software for running its noise reduction program. That's a far cry from implementing such a system in-ear. The situation is changing rapidly, however, and the effects will be revolutionary.

ML Noise Reduction Goes Portable

A company called Whisper was the first to apply machine learning (ML) deep neural networks to the hearing device case, launching its first product

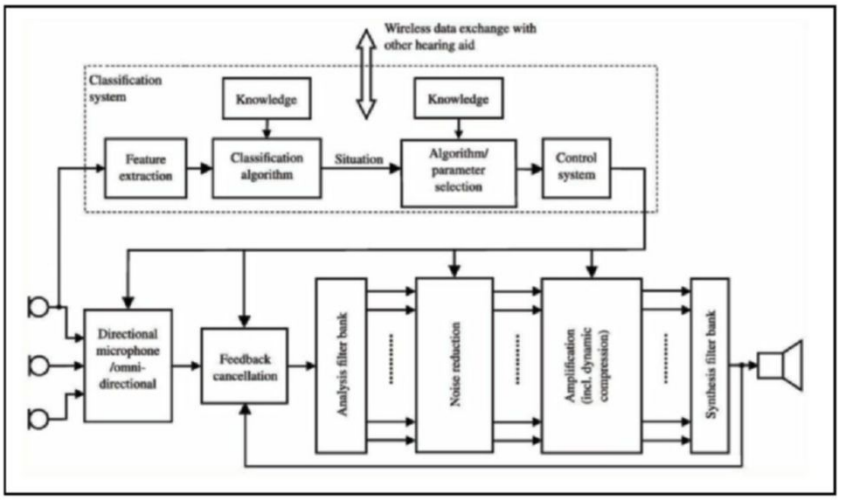

Figure 1: Block diagram of a typical hearing aid processor [2]

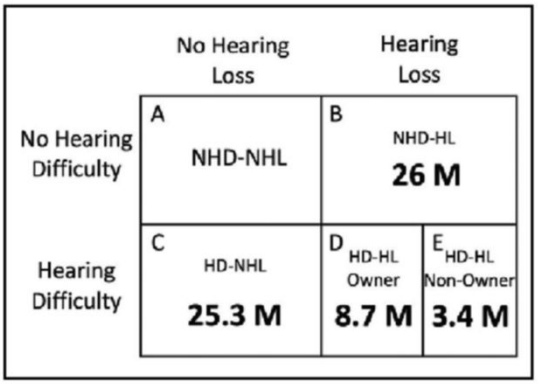

Figure 2: The estimated US adult population with self-reported hearing difficulty and no audiometric hearing loss (box C) is shown here.

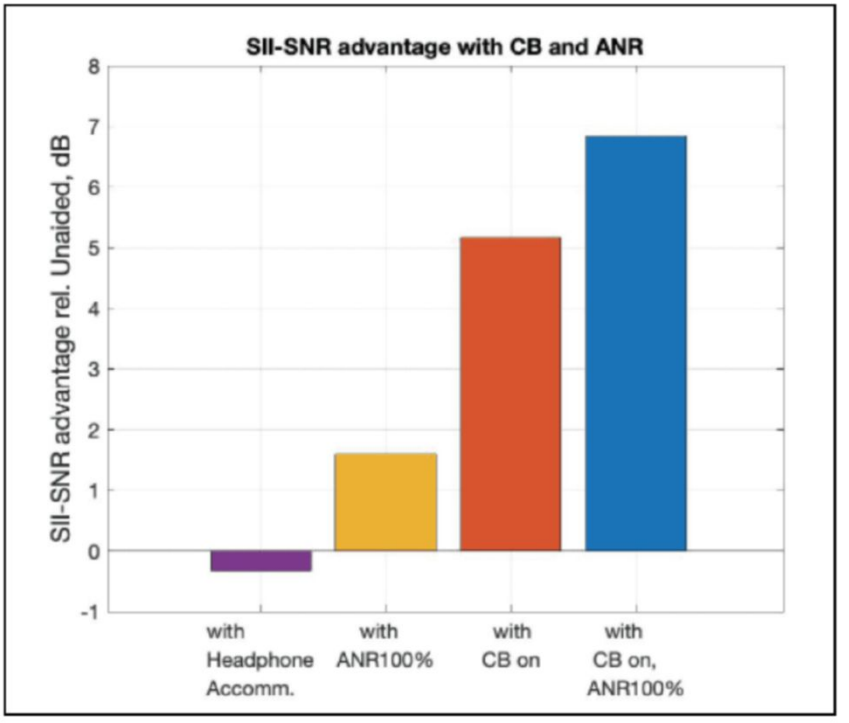

Figure 3: Hearing SNR improvement is illustrated with the AirPods Pro hearing features turned on.

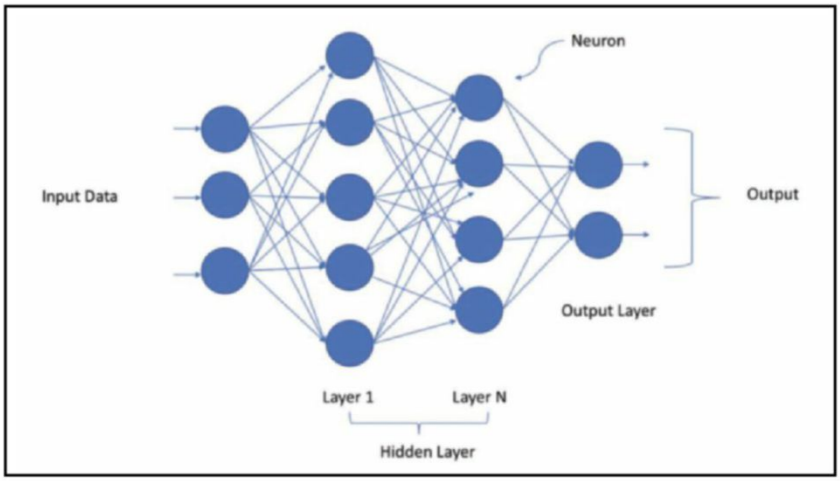

Figure 4: This shows a deep neural network with N hidden layers [7].

Figure 5: A Whisper hearing aid uses a separate "Brain," about the size of a mobile phone and using a similar processor. (Image Source: Whisper)

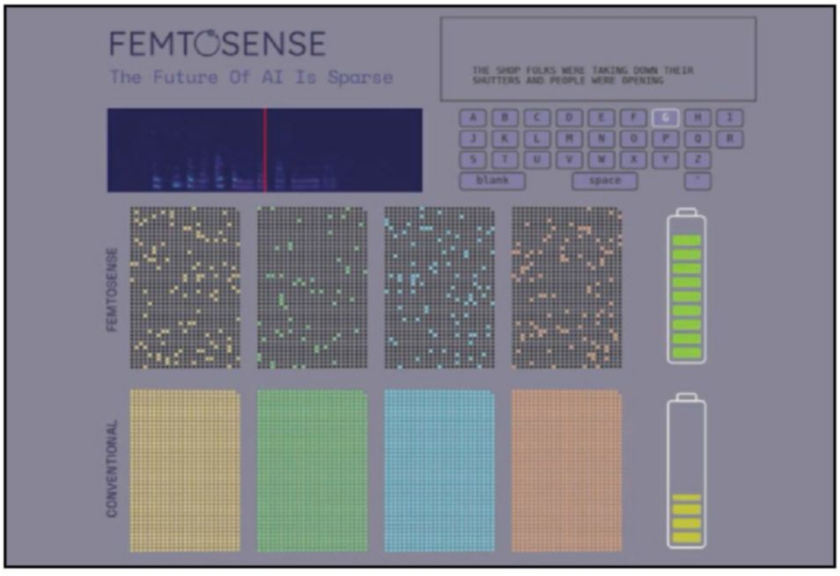

Figure 6: This is an illustration of a Femtosense processor showing active nodes at a given moment. (Image Source: Femtosense)

in 2020. It was still not possible to run the neural network in-ear, so the Whisper system uses a separate "Brain," about the size of a mobile phone and using a similar processor. The hearing aids themselves are of a classic design and work without the Brain. When the Brain is turned on, received audio is passed from the hearing aids to the Brain, de-noised, and returned at very low latency. Not only does the use of an external device allow enough processing power to run the neural network locally, it is also updatable. As of this writing, Whisper has delivered its fifth major software update to its customers (**Figure 5**). It is not likely that such a system would gain mass-market acceptance, however. Consumers will balk unless the entire system is in-ear and provides noticeable benefit. This is where it gets interesting.

A Neural Network in Your Ear

With today's technology, typical processors that can be incorporated into a hearable do not have the resources to run a deep neural network, especially considering limitations on allowed power consumption. That is poised to change as companies develop processors specifically designed to address this need. Greenwaves Technologies is sampling its GAP9 processor with a nine-core RISC-V compute cluster that they claim "is perfectly adapted to handling combinations of neural network and digital signal processing tasks delivering programmable compute power at extreme energy efficiency" [8]. Greenwaves has been demonstrating noise reduction and other applications on the GAP9, and has reserved a suite at CES.

Femtosense has taken a different approach by designing a neural network in hardware. It uses what they call a "sparse" architecture, activating only those nodes in the network that provide useful results at any given moment. They claim significantly reduced power consumption as a result, since only a fraction of the notes are turned on at any given time (**Figure 6**). Although the Femtosense chip could be implemented as a stand-alone co-processor, it is more likely to find a home as an IP block within a multifunction SoC for the sake of efficiency. Having already demonstrated its de-noising algorithm running on a development board at CES 2022, it will be most interesting to see its progress at the 2023 event. Solutions such as these will bring hearables one step closer to providing broadcast quality denoising in-ear, potentially revolutionizing how consumer devices address the cocktail party problem without amplification.

Evolving Toward Smart Hearables

Hearable devices have been following the same trajectory as mobile phones, in three stages. The

first stage was the introduction of mobile phones whose sole purpose was to originate and receive calls. Next came feature phones with more advanced functions, such as email and web browsing preprogrammed. Finally, with the release of the original iPhone, came the app store ecosystem. By democratizing access to what amounted to a pocket-sized computer, innovation exploded. Today, we are at the "feature hearables" stage, but the same progression is about to happen, with important consequences in the hearing space.

One company working to bring a true operating system to hearables is Bragi, which pioneered true wireless devices in the previous decade and more recently pivoted solely toward developing its OS. Today, Bragi offers a range of apps that can be pre-configured at the factory or sold afterward through an app store. Though the complexity of the apps is limited by the processing power of the supported SoCs from Airoha and WuQi, a glimpse of the future is seen by the inclusion of Mimi Hearing Technologies' app for personalizing streaming audio playback to one's individual hearing profile (**Figure 7**).

While Bragi is working with existing SoC suppliers, Sonical is taking a different approach by developing its own processor (**Figure 8**). Optimized to run its OS, Sonical claims it will "provide the world's first ear computer that has the capabilities and performance for next-generation ear worn products." In a webinar hosted by Danish Sound cluster, Gary Spittle, founder and CEO, promised that the "world's first ear computer platform" will launch at CES, and that "many plugins are already available."

A New Wave of Innovation

By providing a platform for software developers to innovate without having to create corresponding hardware, smartphones have opened up whole new worlds. Think about all the smartphone apps you use and imagine a similar ecosystem running natively in your true wireless earphones.

Coming back to the cocktail party problem, there are companies working to mimic the brain's ability to concentrate on and filter the desired speaker even more closely. The smart hearable ecosystem will provide the means to bring these innovations to practical use.

In the same webinar at which Sonical teased its CES presence, Neurotech software company Segotia discussed research performed at KU Leuven (**Figure 9**) as basis of its own work to detect the specific voice being attended to by measuring EEG signals [9]. EEG sensor developer AAVAA also sees a future in hearables, highlighting that its "end-to-end solution fuses our advancements in attention and audio software, sensors, hardware, and design" in a modular platform (**Figure 10**). It is not hard to imagine how beamforming mics and ML-based denoising algorithms could be enhanced by knowing which person the user is attempting to hear.

Future hearable apps could detect the microphone and sensor capability of a particular device and configure itself for optimum performance. Much like picking out a smartphone based on a mix of hardware features and performance (think screen resolution, camera performance, and sound quality),

Figure 7: This illustration shows a Bragi in-ear app ecosystem. (Image Source: Bragi)

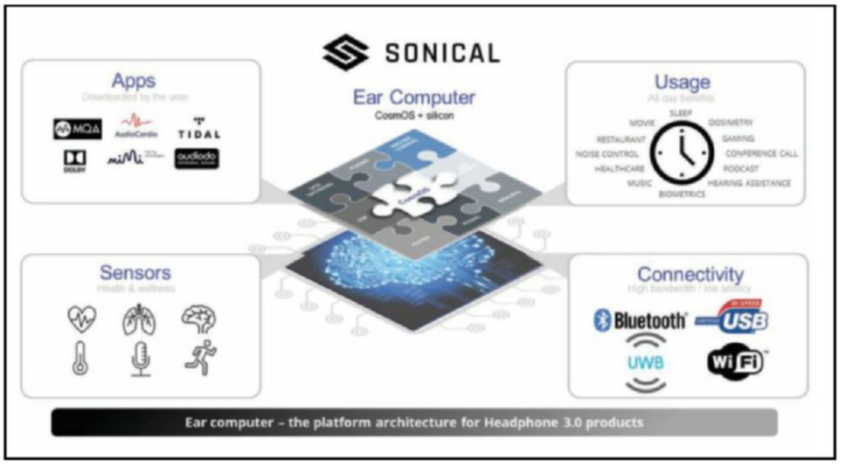

Figure 8: Sonical's hearable processor and app ecosystem wants to provide 'the world's first ear computer." (Image Source: Sonical)

About the Author

Prior to founding AuraFuturity, a marketing consulting company focusing on in-ear and hearing, Andrew Bellavia had experience in international sales, marketing, product management, and general management. Audio has been both an abiding interest and a market he served professionally in these roles. Andrew has been deeply embedded in the hearables space since the beginning and is recognized as a thought leader in the convergence of hearables and hearing health. He has also been a strong advocate for hearing care innovation and accessibility. Andrew's home base at CES will be the ALTI suite, Lido 3101B.

a consumer will be able to choose a hearable with optimum performance for their primary use cases, knowing they will still work with a wide range of other apps. It all amounts to a playground for innovation that will yield exciting results for hearing enhancement.

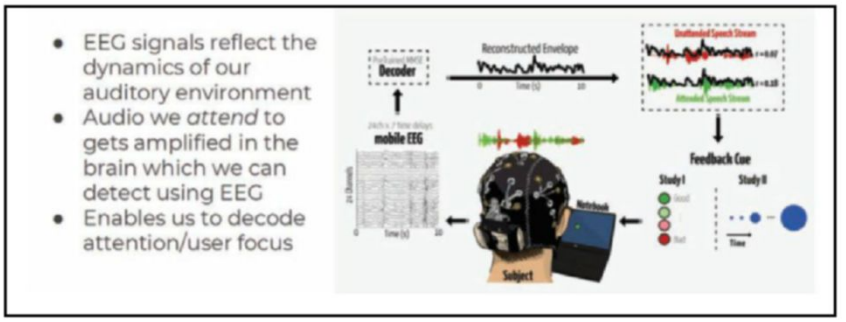
EEG signals reflect the dynamics of our auditory environment
Audio we attend to gets amplified in the brain which we can detect using EEG
Enables us to decode attention/user focus
Decoder
Reconstructed Envelope
mobile EEG
Feedback Cue
Study I
Study II
Subject

Figure 9: This Segotia slide describes the KU Leuven study cited in text. (Image Source: Segotia)

On-the-edge ML & SP processing
Attention and interactivity commands generated
Custom personalized applications
Discreet brain and biosensors

Figure 10: AAVAA's EEG sensor implementation is shown. (Image Source: AAVAA)

Regardless of Approach, Consumers Win

Even with techniques such as de-noising speech and beamforming mics steered by detecting which sound one is attending to, selective amplification will not be totally banished from hearing enhancement devices. But it looks increasingly likely that even amplification will be available from third-party app developers.

Already Mimi Hearing Technologies' app for personalizing streaming audio, an unregulated hearing feature, is becoming available on both the Bragi and the Sonical platforms. It is well within the capability of app developers to offer a similar program for the microphones, essentially converting a smart wireless earphone into an OTC hearing aid after the fact.

It is difficult to imagine how the FDA will regulate earphones, which do not themselves provide amplification even if a percentage of consumers will buy an app from a third party afterward, and non-amplifying hearing features are not regulated at all. The newly released OTC hearing aid rule will be left by the wayside should these promising developments become reality.

One thing is clear. While the pace of developments may in time give regulators and some other stakeholders fits, the consumer stands to benefit from all the innovation coming to hearables. With the ability to "try before you buy" a variety of hearing apps and load them into mass-market consumer earphones, more people will find an effective solution for their hearing and lifestyle, with less stigma. That can only be a good thing.

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The Audio Foundry Automotive Audio Development Platform

By
Adam Levenson

This article details the possibilities of the Audio Foundry platform for automotive audio prototyping and to support the development of audio experiences in vehicles with powerful design tools and services. The result of a collaboration between Tymphany and DSP Concepts.

It's rare to find genuine innovation in the audio tech field, and even rarer to get an opportunity to contribute to a groundbreaking project. Back in June 2022, at the Audio Engineering Society (AES) Automotive Audio Conference in Dearborn, MI, surrounded by gleaming luxury demo cars, DSP Concepts and Tymphany set up a live demonstration in the corner of the room. Automotive OEMs and Tier 1s began lining up to step inside the brushed aluminum acoustic hulk known as the Rig, powered by Audio Weaver and displayed on a big screen perched in front of the Rig's windshield.

As Tymphany's Matt Marchese put it, "You could see it, sitting in the Rig with these OEMs and them just having this OMG moment. It was an eye opening moment, even for us." Soon after, the Audio Foundry was born.

To understand the vision and the scope of this totally new offering for the automotive industry, I sat down with the Audio Foundry team, which includes Matt Marchese (Automotive Business Unit Leader, Tymphany), Evan Bunner (Engineering Services Team Leader, Tymphany), John Whitecar (VP of Product Management, DSP Concepts), and Steve Ernst (Head of Automotive Business Development, DSP Concepts).

Naturally, each member of this team has a particular vision of the project, as with different pieces of the puzzle. John Whitecar calls Audio Foundry "a way to accelerate audio development." Steve Ernst agrees with this fundamental definition but expands on it calling it "a place, specialized equipment, and services for OEMs to come to and accelerate their audio development."

Matt Marchese provided the more practical vision. "The Audio Foundry is also synonymous with the partnership and collaboration between DSP Concepts and Tymphany. It's also synonymous with the space, and soon to be spaces, available at a global level for Tier 1s, OEMs, and technology providers."

A Changing Market

Naturally, we asked the team to explain the market conditions that inspired the creation of the Audio Foundry and what called for the solution. "I always like to say that this is an unprecedented time in the industry where the technology in the vehicles is changing, as are the vehicles themselves," Marchese responds.

Whitecar says, "I think there are two key aspects: First, the complexities of the audio experiences in a vehicle are changing rapidly. For 20 years, audio has been pretty staid in cars but the explosion in electric vehicles (EVs) has really opened up a number of new use cases, a number of audio experiences. Plus concurrent use cases drive all of these audio functions at the same time: playback experiences, voice experiences, emergency vehicle detection, AVAS... All of that is coming together all at one time. The other aspect of this is the fact that the

compute power available is so vast now, that it's very cost effective to implement these features. People have known about these features for years, but they were too costly. The capability of silicon, when you've got multiple DSP cores integrated into a part, suddenly opened up a whole new world of what you can get done.

These developments are also fairly recent. I would say probably the last three years. But when you look at the compute power perspective, the Qualcomm Snapdragons with embedded Hexagons, that's kind of the first platform that really enabled all this. It allows moving audio processing from an external DSP into an embedded processor. Obviously Tesla came out with EVs over 10 years ago, but it's only recently that the market has really started to pick up. You look at what's going on in China. You look at every major OEM that has now pivoted and shifted to EV manufacturing. The whole industry is moving right now."

Steve Ernst adds, "What's also driving this are requirements. There's a use case expansion, there's this bandwidth opportunity or processing power capability, but there are also now requirements that are driving it. Many of these are not elective. When you look at these new groups that are leveraging audio, it is out of a pure legality perspective: they have to do it. There are international standards related to EVs. Because now, for example, you need to create external sound in order to alert pedestrians. Autonomous would be the other driving force: when autonomous is engaged, there are safety alerts and things that have to take place in order to make sure that person stays engaged and understands the state of the vehicle when it's driving in autonomous mode. So it's beyond, call it fun and entertainment, it's now moved into a whole different realm of requirements in order for the vehicle to be drivable."

And Marchese confirms, "We've heard from almost every OEM that they're looking to create unique experiences in the vehicle. They're looking for more and more ways to differentiate and this idea of experiences in the vehicle has become a main driver and audio helps support that."

The Automotive Audio Challenges

This in summary explains the requirements and the market conditions. But this project directly reflects the need to find new solutions for the challenges the OEMs and Tier 1s are facing when building audio systems. The experienced team behind Audio Foundry is in a unique position to understand these new audio experiences in vehicles, and consequently illustrate what problems and solutions they proposed to solve.

According to Ernst, "The OEMs and Tier 1s have not changed the way they've done business for the last 25 years, 50 years. That is clearly the challenge for them because we promote this concept of the modern sourcing model, which is completely juxtaposed against this idea of the black box. There's so much at stake now, and OEMs want more control and ownership, but it changes how they do business and where the skill sets are, and that's the biggest hurdle for them."

"When we talk to the OEMs, even when they emotionally make this determination to use Audio Weaver—as an example—it comes back to, 'How do I do that?' And, 'I know I need it, I know I want it, and I gotta have it, but I don't know how to do it, and I don't have the facility, talent, or resources to do it.' Part of the Audio Foundry's value proposition is our ability to support the OEMs no matter where they are in terms of readiness to achieve their audio goals. This is a discovery platform for them now to figure out how they're going to manage this new

The Audio Foundry platform offers purpose-built labs for automotive audio prototyping, offering listening rooms, measurement rooms, studio rooms, and anything needed to support engineers with powerful design tools and services.

A fundamental part of the Audio Foundry platform, the Rig is an SUV-sized audio development tool used for creating new acoustic solutions, layouts, and proof of concepts. It offers an easily adjustable, modifiable acoustic environment with 24+ locations for speaker placement with replaceable panels, delivered with services and maintenance packages.

Steve Ernst, Head of Automotive Business Development, joined DSP Concepts in January 2021 in Santa Clara, CA. He has a 30-year sales and marketing career in automotive audio, having worked as business unit leader in companies such as Harman International, stationed in the US, Japan, and South Korea.

Matt Marchese, Automotive Business Unit Leader at Tymphany, is based in San Francisco, CA. Matt began his career in automotive with in-vehicle electronics at Ford/Visteon, worked at CARLAB, an automotive product planning consulting firm for nearly every global OEM, and previously worked for THX, supporting audio/video certification and licensing.

development cycle and purchasing strategy."

Whitecar added that, "The automotive industry has been very hardware focused. Traditionally, they are hardware mechanical people putting stuff into cars. So that's the other big paradigm shift: The industry is moving very quickly to a software model. And so, back to Steve's point, they don't know how to do it because it's a completely different skill set. It's going from: 'I can build you a box that does these two features,' to 'I've got to get this feature into my software environment onto a main SoC, or our main module that's controlling everything.' So you're changing the skill sets, you're also changing how they organize teams."

Whitecar continues, "It used to be, 'I have a team that builds this box, and I've been building this box for every car for the last 20 years, and my team knows everything there is to know about this box.' And now that box is gone. So now it's like 'How do I get this done?' 'My normal team is broken, my approach is broken, my processes are broken.' So they have to figure out how to pivot and move in to this heavy software domain that leverages major hardware components. Now they have to get teams to work together. The NVH (Noise, vibration, and harshness) team has to talk to the audio team, has to talk to the infotainment team, and the voice team. They all now have to work together to build these products."

Who Is It For?

So, who is asking these questions and seeking the new solutions? What is the role of these people at the OEMs and Tier 1s?

According to Ernst, "It's not the VP, it's that layer below. The person who owns the department that makes the actual deliverable. They're getting the assignment from the VP to say, 'Hey, by the way, we're a software company now. Deliver everything to me as software, not hardware.' So it's that layer below management that has to figure out what that means and has to define how it's going to happen."

Marchese says, "I also think about the re-empowerment of the automotive audio engineer. There's been this heavy reliance on the Tier 1 to bring forward the solutions, bring forward the innovation, and I think that's where you can now see the shift. Audio engineers are now wanting to take control of their own destiny and do this work themselves. I think the Audio Foundry really helps support and gives them a venue to explore, experiment, and develop."

"And, especially for some of the startup companies, where there's this idea of creating more consumer centric approaches like having removable

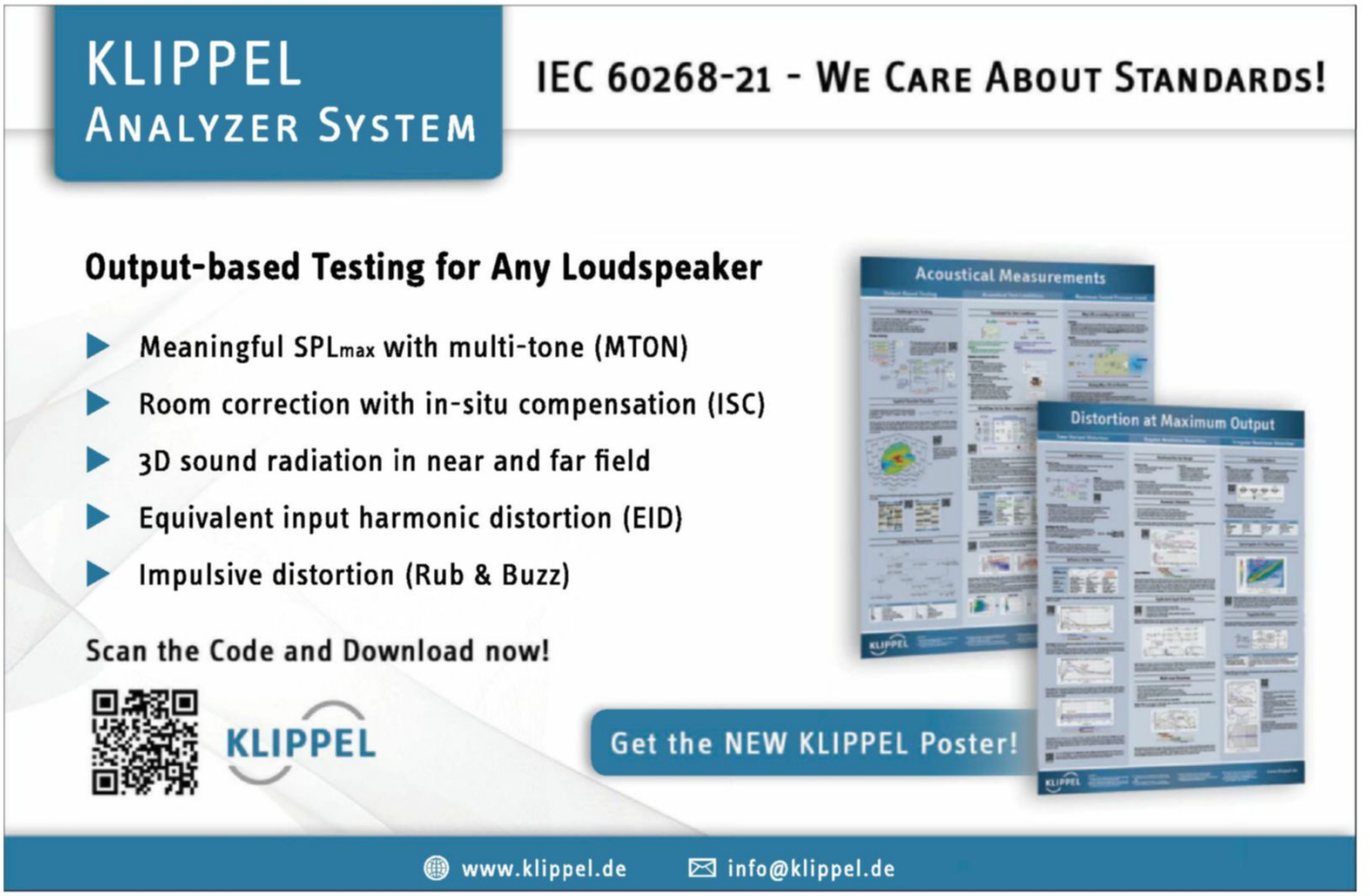
speakers or having headphone sets. They're basically creating a whole different experience in the audio system, which requires hardware aspects, transducer aspects, system aspects, and then there are software aspects that all have to be proven out and put it into program. So Audio Foundry gives managers a way to work on these concepts and present them to the decision makers. I think Steve's right on: I think it's that level at which somebody owns the problem of having to deliver the features, and they may not have the headcount or the facilities to actually pull it off," explains Whitecar.

Evan Bunner, Engineering Services Leader, Tymphany, based in San Rafael, CA, is a proven Automotive Design Engineer, experienced in Product Development and Vehicle Engineering. Evan has been developing automotive components since the beginning of his career in 2013, when he started by developing composite structures for racing vehicles and supported NVH research activities for Honda.

The Audio Foundry's Backstory

So, how does a partnership like this get started? According to Whitecar, Marchese had the vision. Tymphany was working on a proof of concept and had already developed an SUV-sized automotive audio development environment for its own internal uses. But it became clear that there was some synergy between the hardware and the software, and the possibility of building out experiences and bringing a platform and a partnership together that OEMs and Tier 1s could use. So Marchese called DSP Concepts and that was the genesis of it.

"I've known John for some time, since I was working in the consumer side of the business. In getting feedback from Tymphany's own individual pursuits and noting how often Audio Weaver would become part of those discussions, it just became apparent that, within the development realm, we didn't have the software side to complete a platform," Marchese explains "The funny thing is that we spent all of the summer of 2020 talking to software companies about how hardware and software could go together. And honestly, we had a very difficult time finding a partnership that really stuck or had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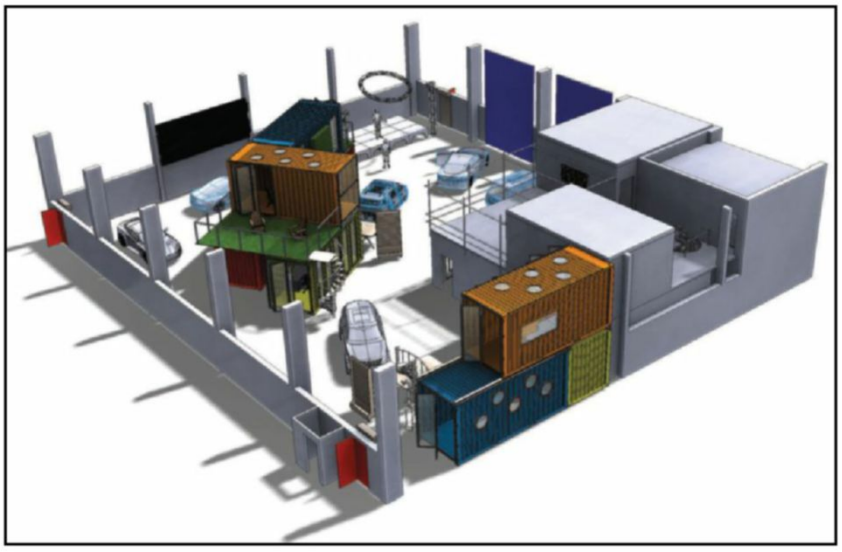
John Whitecar, VP of Product Management, Automotive at DSP Concepts, from Stanford, CA, is a world-class automotive audio technology expert with c-suite and VP level engineering experience at Tesla Motors, Ford, Visteon, and Texas Instruments. He holds 35 patents covering audio, acoustic, and receiver signal processing technologies.

stickiness to it. This model, this early engagement services model is not new to Tymphany. We've been doing this on the consumer and pro audio side, and we've been trying to find a unique approach into automotive. So with our pre-existing partnership with DSP Concepts, and knowing what DSP Concepts means in the industry, it was a no-brainer to reach out and see if there was an opportunity to expand the partnership into automotive."

Marchese also points to the AES automotive conference in Michigan as the pivotal point, when the two companies demonstrated the vision of hardware and software working together. The positive reactions from the attendees showed that the companies had something there (the full report is available in *audioXpress*, September 2022).

"We had the exact right demographic including Tier 1s and OEMs sitting in the Rig with Audio Weaver.



The Audio Foundry Makerspace. Fully equipped, state-of-the-art facilities located in San Rafael, CA, and Wales, UK, which can be leased and staffed.

It was like, you know, these two things coming together to really accelerate the development. You could see it, sitting in the Rig with these OEMs and them starting to discuss different scenarios in which they could see how this could help improve their timing, improve their costs, help them to improve the way they innovate. It was an eye-opening moment, even for us," recalls Marchese.

The Audio Foundry Advantage

As Evan Bunner explains it, the Audio Foundry platform is intended to be a one-stop-shop for proving new ideas, trying something new, and finding the next game-changing technology in the automotive industry. "The idea is to have all of the ingredients available to make that possible for a wide range of OEMs. At the core, you'll have a workshop to do hardware stuff, and a platform to implement software innovations."

"There's this really classic process for designing products, any product. It goes back to this classic V diagram where you have requirements on one side of the V and system validation at the side of the V. You start with the big objectives, work your way down to the little features and pieces of code and individual components, and then build the system back up on the validation side. So you start big with concepts and ideas, and really, the upper left hand corner is fuzzy things such as a new experience that an OEM wants to provide to the customer. It starts very vague. It has high-level requirements associated with it. So the Audio Foundry workflow is to get those things done quickly," explains Whitecar

"To have a hardware platform and a software platform that lets you really spin through ideas quickly. Then you can start refining those ideas, start pushing into hardware, pushing software to specific targets. So you can take this concept that's fuzzy, done at the high level, now you can bring it back into the Rig and test it with closer to production hardware, with a software environment, and push your way down the V towards productization. And then you can actually provide mechanisms for validating what you set out to do as you get closer to the real products. The whole idea is that each step is accelerated because you've abstracted a lot of the low-level details out in the process, which is usually what slows you down. If you think about a typical approach, if an OEM wants to put an audio concept into a car today, they have to tear up a car, they have to cut holes in the car, they have to redo trim panels, they have to find hardware to run things on. It's a very time consuming and arduous process," Whitecar continues.

Another key aspect, as Ernst reminds us, is the difficulty in getting access to the actual car that

an audio project is intended to fit. "It's not easy to have access to the car. That's no problem for audio engineers working for an OEM, but it's not easy for all others to get access to a vehicle, especially for the period of time that they're going to need to fulfill all of the requirements and experiments. Especially if they want to cut and drill into a prototype car."

It is an experience that Whitecar could confirm happened many times over his career. "Early prototypes are rare resources, and so you have less flexibility. So typically what OEMs will do is they'll take an existing car, a production car that's already on the market, and maybe it's been used, and then you start cutting into that.

"It's very rare to have a hot off-the-line, early prototype, and then start to hack at it. Because that's a very expensive car and there's a lot of people who want to get time with that vehicle. So you have to work offline. And Audio Foundry is even more of an ideal offline solution because it's a clean sheet of paper. It's a 3D sandbox. You can put anything anywhere, and really think through experiences, and not focus so much on how do to get it done. It's more about making it possible to achieve what you want to get done," adds Whitecar.

The first demonstration of Tymphany's and DSP Concepts' environment for automotive audio development during the Audio Engineering Society's 2022 International Automotive Audio Conference in Dearborn, MI.

Hands-On Platform

In addition to offering a hardware and software platform and a makerspace, the Audio Foundry provides support from the DSP Concepts and Tymphany teams. I asked them to describe some of the ways to work with the Audio Foundry.

Ernst points out that OEMs and Tier 1s can start by just commissioning a design-of-experiment for the Foundry team to put together, measure it, and report. "It can start at that level, where it's called a professional service, and then from that point, it can expand if they want to be involved. So it could be that they want to conduct a DOE, make changes, have this

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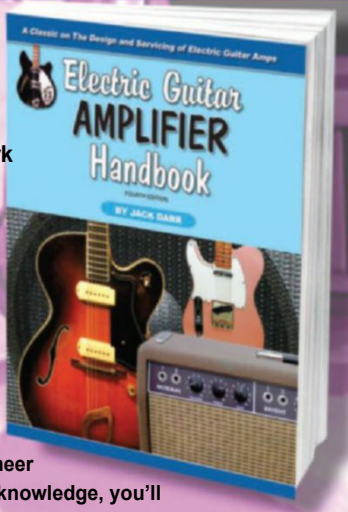

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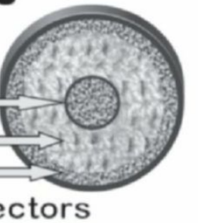

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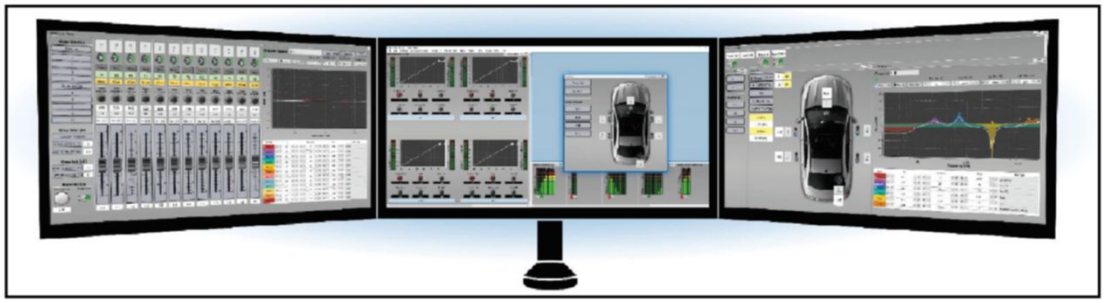

Audio Weaver offers AWE Tune for system tuning of the entire audio system in vehicles, including modules to measure the acoustical fingerprint of a vehicle for all seats, a complete system for filter design and optimization, and real-time sound tuning and listening test tool.

setup, create the hardware, and the Foundry team will work with the client as a part of the project, and then collectively create that report. Another option is that the customer can lease the Audio Foundry makerspace that's fully equipped with the Rig, listening/measurement/studio rooms, workbenches, 3D printers, and more. That enables the customer to do the work themselves with expert Foundry team support. A third option is that the Foundry can build a Rig and deliver it to the customer's facility with training and a services package."

"The Audio Foundry team can help support the collaboration or development of a new system or architecture or proof of concept," Marchese confirms. "And as Steve also said, the client can do that independently in the space and utilize the tools and the environment to do that. The magic is in the space itself, this will be the first of its kind, this does not exist in the industry today. It's a space that's 100% dedicated to automotive audio development. An unique space for OEMs and Tier 1s to go, to work out problems and explore solutions."

"The space is going to mean different things to different car companies. I would love for car companies to look at this, even the most developed ones, as an extension of their current capabilities. Many car companies, older car companies, have very developed audio engineering labs, but to the points that we've made throughout this discussion, those don't include what the Audio Foundry is providing. This may not be their 24/7, 52-week place, but Audio Foundry can be viewed as an extension of their workplace, to get benefits that they're not going to get within their own four walls," Ernst reinforces.

Bunner also believes that the Audio Foundry platform will have different resonances with different companies. "It will be a place to experiment for the bigger companies, but for the startups, it could be their day to day workshop."

"Some of the smaller car companies don't even have a place to tune their car. We know that it's a different story for the big car companies: They have plenty of places to tune and they have their supply base, but it's not that way with the small guys," Ernst says.

Having extensive experience in supporting engineering teams and Tier 1s in audio systems for the automotive industry, Tymphany and DSP Concepts understand how such purpose-built labs for automotive audio prototyping can become a valuable proposition for early concept validation, accelerate development, increase design flexibility and facilitate in-vehicle concept presentations.

The Audio Foundry platform offers fully equipped workspaces, the Rig development sandbox, onsite audio experts, and the power of Audio Weaver for audio development, with integrated debugging for hardware and software, and a multifaceted tuning toolkit.

As Whitecar summarizes, for big and small companies, Audio Foundry becomes a perfect facility to do all kinds of experiments. For more information, visit https://audiofoundry.dspconcepts.com.

About the Author

Adam Levenson's 30-year audio career has evolved from sound effects design, to business development, to marketing with roles as Director of Audio at Activision, GM at CRI Middleware, VP of Product Marketing at Waves, and now Head of Marketing at DSP Concepts.

Tymphany is a leading company in the design and manufacturing of audio systems, headphones, acoustic modules, and drivers. With more than 90 years of audio design, development and manufacturing expertise, Tymphany is widely regarded as having the strongest portfolio of technology and design experience in the loudspeaker industry. Tymphany manufactures OEM/ODM loudspeakers for many of the world's leading audio brands and also develops and sells transducers under the Peerless brand.

DSP Concepts is a leading embedded audio technology supplier and creator of Audio Weaver, the audio development platform that makes audio innovation easy. DSP Concepts equips engineers with real-time workflows to quickly stand up prototypes, collaborate and modify designs across teams, and deploy to the most popular chipsets. from leading semiconductor companies including NXP, Qualcomm, STMicroelectronics, Cadence, Arm, and others. DSP Concepts technologies are embedded in millions of automotive and consumer products, including for automotive brands including BMW, Mercedes Benz, Porsche, Tesla, and many more.

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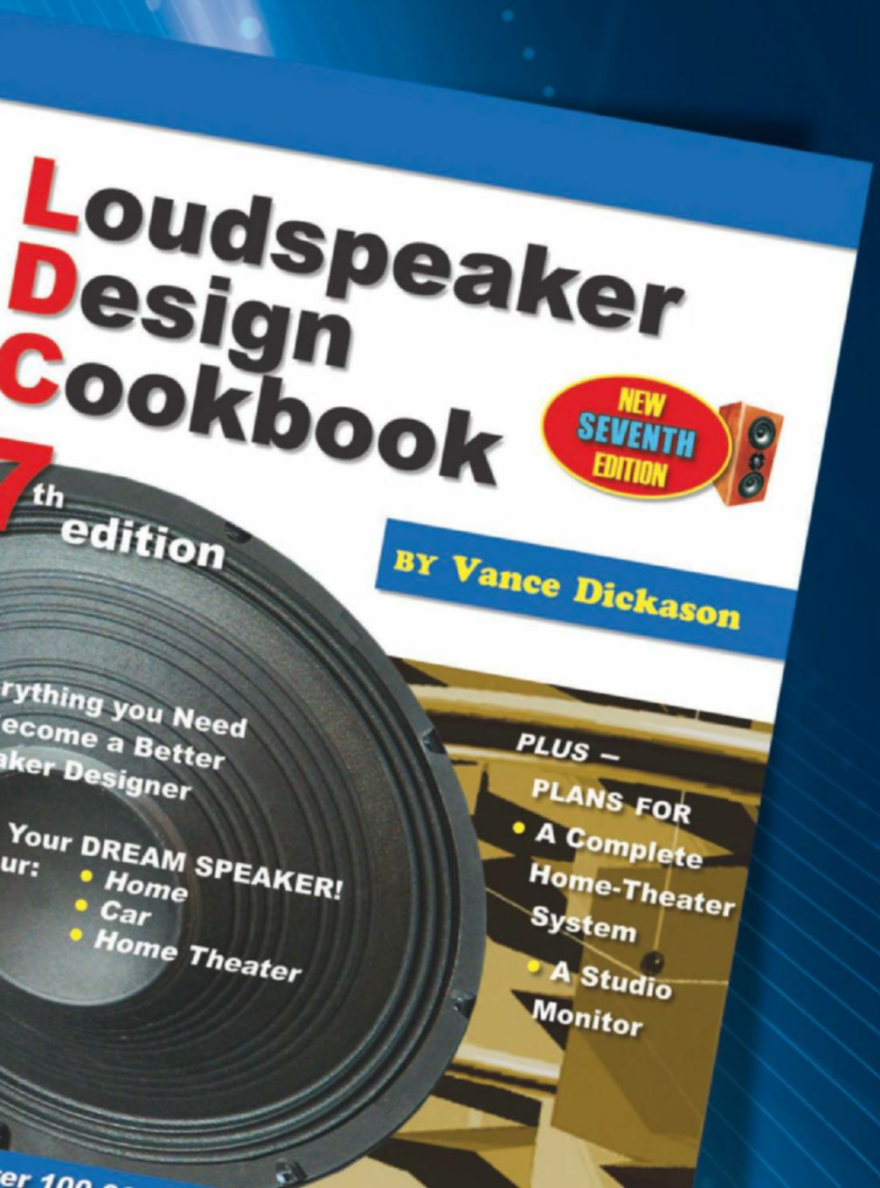

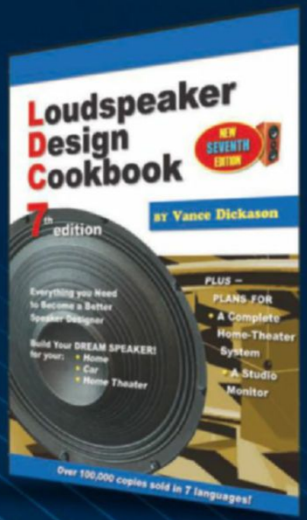

Loudspeaker Driver Displacement Decomposition

This valuable tutorial offers a unique overview on the possibilities of three-dimensional analysis of loudspeaker driver displacement, using Phase Decomposition and Axial-Symmetrical Decomposition techniques.

By
René Christensen

When looking at a frequency response from a loudspeaker driver (magnitude and phase of the pressure in a microphone point), a lot of information can be extracted regarding its behavior. The magnitude response is typically the primary focus, and a flat response here is the desired outcome, whereas the (oftentimes ignored) phase response holds information about the temporal aspects of the loudspeaker and generally of the setup regarding number of drivers and how they combine to the full response. However, the sound pressure is simply a zero-dimensional signal, which does not necessarily tell us the whole story about the three-dimensional aspects of the surface vibration and how it generates the sound. To really dissect where, for example, dips and peaks in the pressure response come from some other tools can help, and two such tools are the Phase Decomposition and the Axial-Symmetrical Decomposition techniques, respectively.

The Phase Decomposition technique [1] as well as the Axial Symmetrical Decomposition [2] have been promoted by Klippel, where they are built into measurement systems and so can be applied to measurements only. I was involved in the development of an add-on package that could take in simulated surface (cone, dust cap, surround) displacement data, and analyze it as it had been measured. But shortly thereafter I developed a simulation setup in COMSOL Multiphysics where the phase decomposition could be carried out directly based on simulation data, and this approach has been used in the analysis and synthesis work for several clients since then.

Phase Decomposition

The first technique relies on knowing the complex pressure in a microphone point and relating the complex displacement across the loudspeaker surface to the pressure, or more specifically to the phase of said pressure. Knowing these quantities, the following displacement components can be established:

$$w_{in}(Q,p,R) = Re^{+}\left(w(Q)e^{-i\phi_{ref}}\right)e^{i\phi_{ref}}$$

$$w_{anti}(Q,p,R) = Re^{-}\left(w(Q)e^{-i\phi_{ref}}\right)e^{i\phi_{ref}}$$

$$w_{in}(Q,p,R) = Im\left(w(Q)e^{-i\phi_{ref}}\right)e^{i\left(\phi_{ref}+\frac{\pi}{2}\right)}$$

with a reference phase of:

$$\phi_{ref} = \arg\left(p(P)\right) + \pi + kR$$

I outlined the details in two separate articles that I wrote on the subject [3] and [4], so let's dive into these equations, starting with the last one first. Looking at the reference phase, the first part is simply the phase of the pressure in the microphone point. This is important, as this is basically the phase that we want to compare with the displacement phase. The middle part indicates a sign shift; this relates to the fact that the pressure in general correlates with the acceleration of the cone and the displacement is proportional to the negative acceleration. The final part is basically a subtraction of the phase related to the path from the integration

point Q to the microphone point P; a subtraction of the irrelevant phase coming from the time delay/transport delay. The total displacement in a point Q on the surface is denoted by w(Q), the respective in-phase, anti-phase, and quadrature components as previously listed. The in-phase displacement is the displacement that is in-phase with the reference phase, which I already explained is the pressure phase taking into account the sign and the distance aspects. This part of the displacement adds to the total pressure. The operator Re+ returns the positive part of the real part. Basically, the equations indicate rotations, projections, and back-rotations according to **Figure 1**, which shows the complex displacement phasor w(Q) and the complex pressure phasor p(Q) along with the component axes.

Similarly, the anti-phase displacement components subtract from the total displacement. And finally, the quadrature displacement component neither adds to nor subtracts from the total displacement, with *Im* being the imaginary part operator. Both the in-phase displacement and the anti-phase ditto have associated in-phase and anti-phase pressures, respectively, whereas the quadrature displacement does not affect the pressure and so has zero associated pressure.

For the Klippel implementation, the pressure is not measured, but instead calculated via the so-called Rayleigh integral:

$$p(P) = -\frac{\omega^2 \rho_0}{2\pi} \int_S w(Q) \frac{e^{-ikR}}{R} dS$$

with the angular frequency ω, the density of the medium ρ_0, the distance from the microphone point to the points Q on the surface under integration, and *S* being the entire surface to be integrated. Using the Rayleigh integral, there is an assumption that the surface is flat and placed in an infinite baffle (**Figure 2**).

The latter requirement fits well with how the typical simulation setup is anyway, but the former is an issue, since most drivers are not flat. Still, the overall analysis and conclusion are typically of value, and a correction can be established before running the actual phase decomposition. These issues were explored in an Audio Engineering Society paper that I co-wrote with Urik Skov [5], but for each case one must evaluate the effect of not fulfilling the "flat" requirement. (One thing to note is that while the varying distance coming from varying depth on the cone is in fact included in the calculations, the dipole aspect of the full Kirchhoff-Helmholtz equation that the Rayleigh integral can be derived from, is not included, but these details are left to reader.)

For the COMSOL Multiphysics implementation, the displacements are found via a laser scanner, which only considers the axial displacement, whereas in a simulation software setup, the user also has access to the radial and normal displacements, which can lead to more accurate results. (In the present work, the simulation software COMSOL Multiphysics has been used throughout.)

Figure 3 shows how a simple total displacement can be decomposed into in-phase, anti-phase, and quadrature displacements.

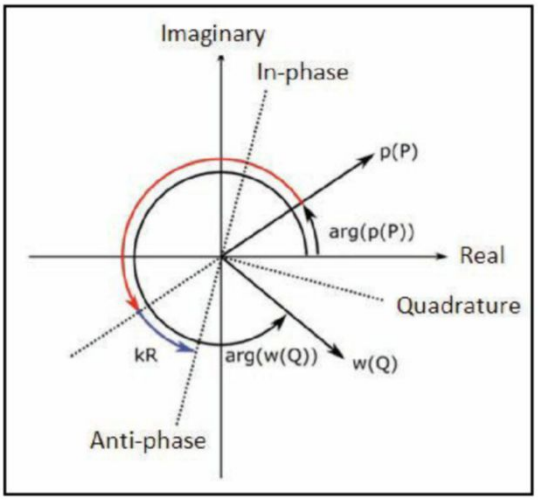

Figure 1: The displacement and pressure phasors are shown here.

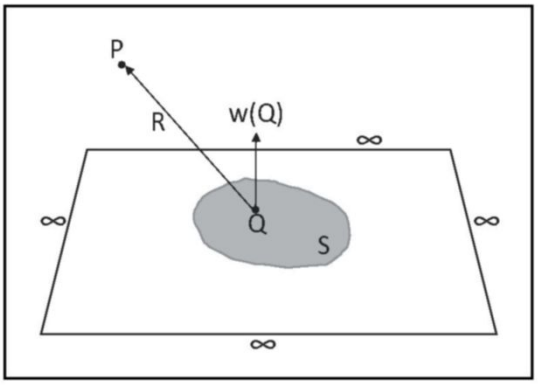

Figure 2: A flat vibrating surface *S* is placed in an infinite baffle. At point *Q* on the surface, the total displacement is *w(Q)*. The distance from point *Q* to a microphone an observation point *P* is denoted *R*.

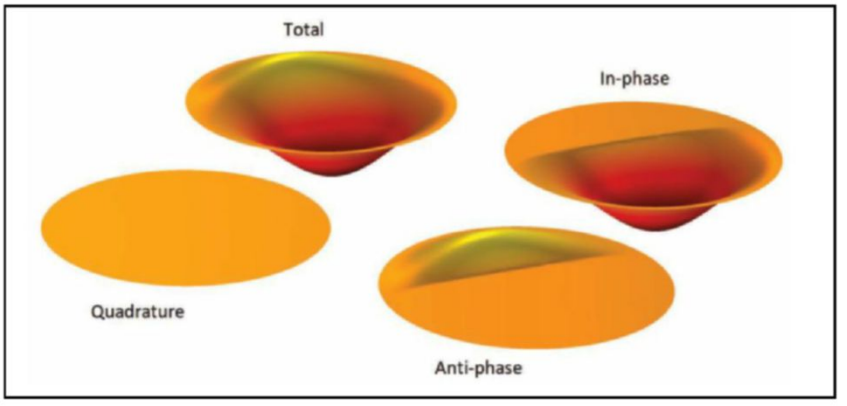

Figure 3: The displacement components for a vibrating disk are shown for an on-axis and far-field observation point.

On the surface the displacement only has two phases—"inward" and "outward"— and so the bigger of the two sections will be the in-phase component driving the pressure, and the smaller section will be in anti-phase, and the quadrature component will be zero.

While the total displacement stays the same, the decomposition will vary depending on the microphone point in question as individual distances to the surface will vary accordingly, and so very different results can be seen off-axis and in the near-field.

For a more practical case, I have used an example from one of my clients, Charalampos Ferekidis (lampos@ferekidis.com), which is shown with his permission. **Figure 4** shows the before (a) and after (b) sound pressure level for an initial driver and the subsequently modified driver. A dip was seen at 1kHz, and its origin was unclear.

A phase decomposition analysis revealed the surround as the culprit, as its contribution was mainly in-phase, meaning driving the resulting sound pressure at that frequency, while the remaining displacement was largely quadrature, and so not adding to the pressure (**Figure 5**). Essentially, the driver is less efficient at radiating sound, since the effective area of radiation was located near the surround.

Other very interesting results can fall out of phase decomposition, and once it is implemented, it will typically be used extensively. The additional components such as Accumulated Acceleration Level (AAL) used within the Klippel environment can also be calculated within COMSOL Multiphysics, and typically with a higher spatial resolution in the simulation setup.

Axial-Symmetrical Decomposition

The axial-symmetrical decomposition is slightly more straightforward when it comes to how it works, but not straightforward to implement. I recently discussed this implementation in the blog post that I wrote for Acculution [6]. This decomposition does not rely on knowing the sound pressure at all, as it works directly on the total displacement alone. The split into components now is between a radial (or more accurately "meridional," but we will stick to "radial" going forward) component and a circumferential component, both to be calculated for each frequency of interest.

The radial component has symmetry around the circumference, and so for example pistonic motion would fall into this category, as would any displacement that can be described as a variation along the radius which is then swept around the

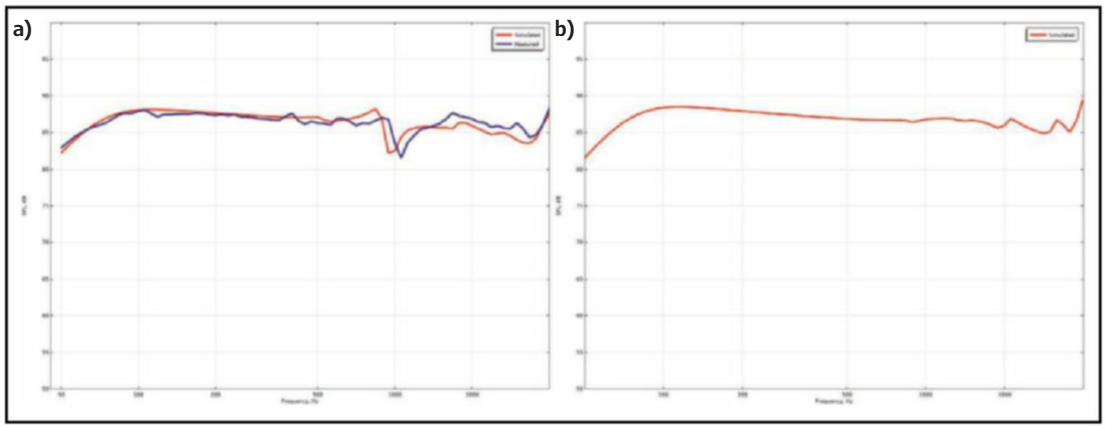

Figure 4: The initial response, both simulated and measured, (a), and the simulated response for the modified driver (b) are shown here.

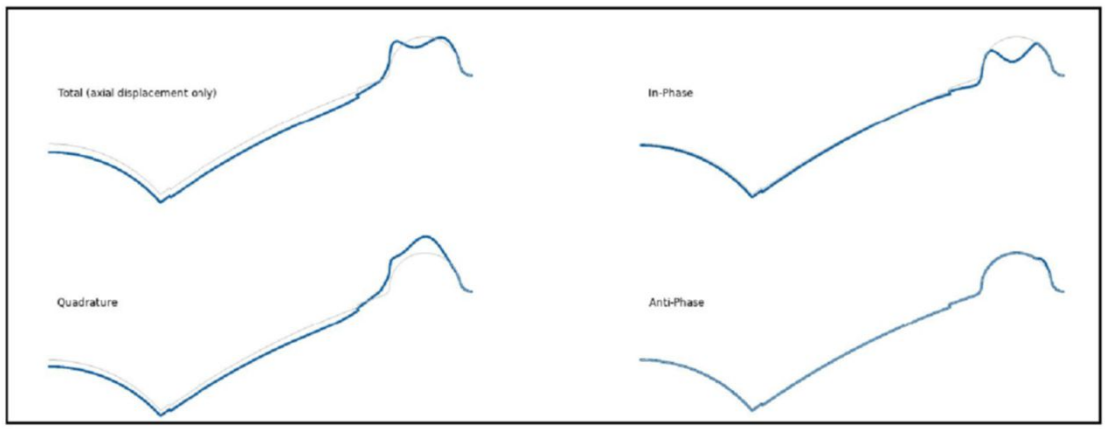

Figure 5: A phase decomposition analysis at 1kHz for a client case revealed the surround as the culprit, as its contribution was mainly in-phase, meaning driving the resulting sound pressure at that frequency, while the remaining displacement was largely quadrature, and so not adding to the pressure.

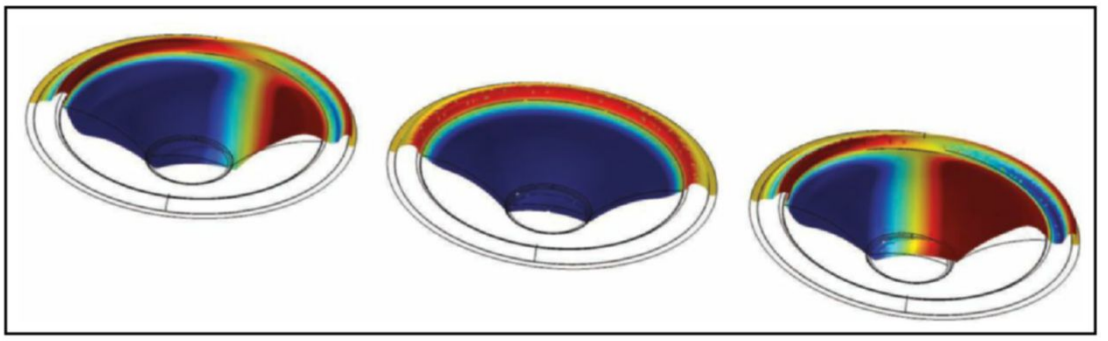

Figure 6: This is an example where the constructed total displacement (left) is combination of a "pistonic" motion (radial displacement in the middle) and a "rocking" motion (circumferential displacement on the right).

axis. The circumferential component is then any part of the total displacement that cannot be described this way. This could stem from modes or asymmetries in the setup. The equation for the radial component is:

$$w_{rad}(r) = \int_0^{2\pi} w_{total}(r,\varphi)\, d\varphi$$

with r being the radius and φ being the circumferential angle. For each r on a continuum, a line integration is carried out along the associated circumferential line, and the resulting displacement varies along the radius only. Knowing this along with the total displacement, the circumferential component can be found as the difference between the two:

$$w_{circ}(r,\varphi) = w_{total}(r,\varphi) - w_{rad}(r)$$

Figure 6 shows an example where the constructed total displacement (left) is combination of a "pistonic" motion (radial component in the middle) and a "rocking" motion (circumferential component on the right). It is seen that the decomposition correctly extracts the individual components. The present decomposition can give insight into modal behavior among other things. Again, having the implementation makes it easy to employ whenever relevant.

Concluding Remarks

It has been demonstrated how a total displacement on a cone surface can be decomposed into different components directly in the simulation software COMSOL Multiphysics, and how this can reveal some of the underlying mechanisms of the driver. The phase decomposition decomposes relative to a complex sound pressure, and certain aspects about limitations of the Rayleigh integral have not been touched upon here but can be found in that AES paper that I co-wrote with Skov [5].

Another consideration is whether to use axial or normal displacements for non-flat surfaces, and that will be discussed in an upcoming publication with Charalampos Ferekidis. The axial-symmetrical decomposition technique decomposes directly on the complex surface displacement but is arguably more difficult to implement.

As the decomposition techniques shown here only consider the surface displacement, whatever is happening behind the driver (spider behavior, acoustic cavity modes) is seemingly not included in the decomposition. But in fact, it is, in the sense that if these effects affect the surface displacement, the decomposition will change accordingly.

Other decomposition techniques such as modal can also be carried out, and it is up to the engineer to figure out which approach will reveal the most insight for the case at hand. In future articles we may look at these alternatives.

About the Author

René Christensen (BSEE, MSc, PhD) has been working with simulations in the loudspeaker and hearing aid industry for several years, and in 2021 he started his own company, Acculution ApS, with a focus on mathematical modelling of many different products, such as transducers. The overall aim is to use academic research results for finding better engineering solutions faster, training engineers in this type of virtual prototyping, and working with students across the world to progress the field.

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[1] W. Klippel and J. Schlechter, "Measurement and Visualization of Loudspeaker Cone Vibration," Klippel GmbH, Dresden, 01277, Germany and Faculty of Computer Science, University of Technology, Dresden, Germany.

[2] "Cone Vibration and Radiation Diagnostics," Application Note to the Klippel R&D System, AN31, April 4, 2012.

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[4] R. Christensen, "Phase Decomposition for Loudspeaker Analysis," GN Resound A/S and Dynaudio A/S, 2016 COMSOL Conference in Munich, https://www.comsol.com/paper/download/356591/christensen_paper.pdf

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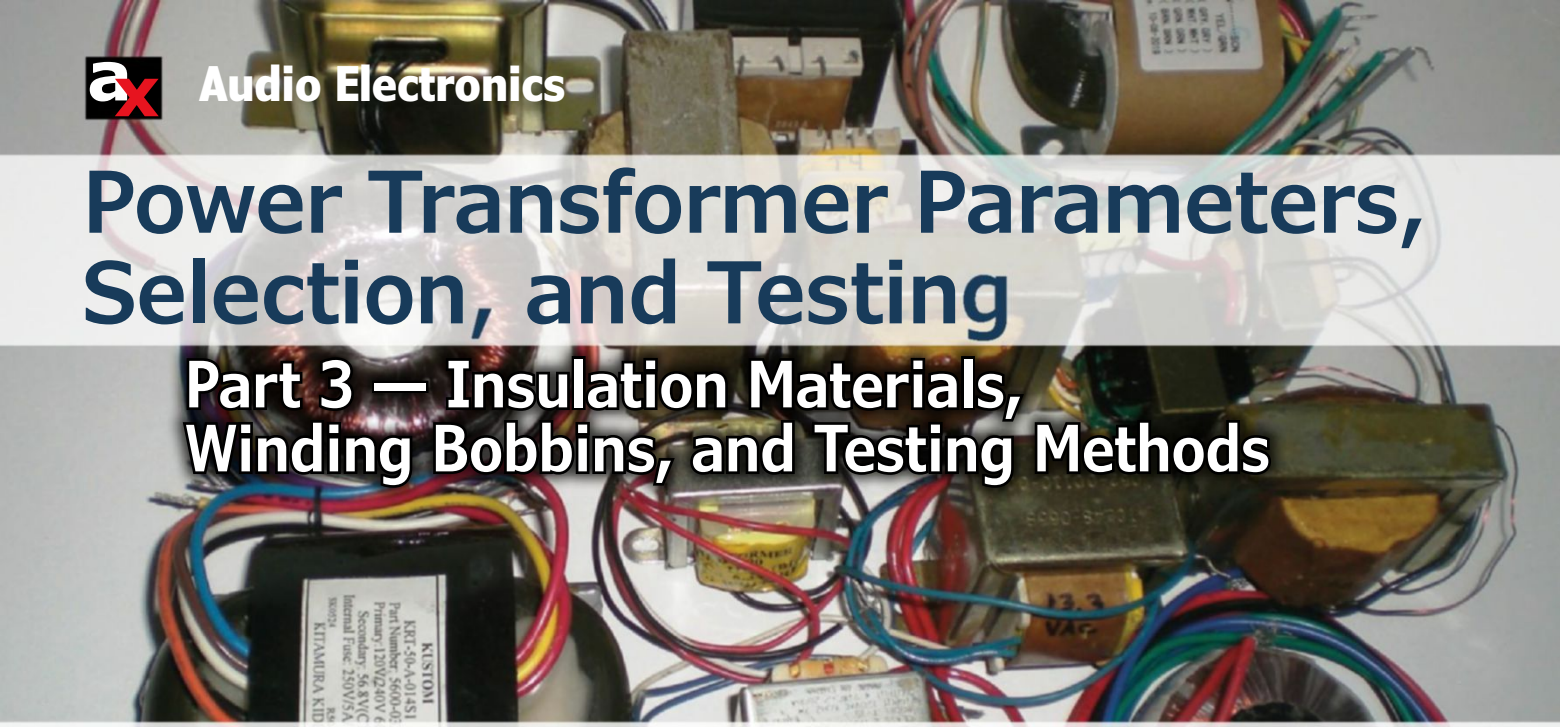
Power Transformer Parameters, Selection, and Testing

Part 3 — Insulation Materials, Winding Bobbins, and Testing Methods



In the previous two articles in this series, Chuck Hansen detailed the history of transformer cores and their various construction methods. This article explores transformer insulation materials and winding bobbins, and testing methods for the insulation system.

By
Chuck Hansen

Commercial transformers are often rated in terms of the maximum operating temperature rise (temperature increase above ambient) according to a letter-class system: A, B, F, or H. These letter codes are arranged in order of the lowest temperature rise to the highest:

- Class A: No more than 55°C winding temperature rise, at 40°C maximum ambient air temperature
- Class B: No more than 80°C winding temperature rise, at 40°C maximum ambient air temperature
- Class F: No more than 115°C winding temperature rise, at 40°C maximum ambient air temperature
- Class H: No more than 150°C winding temperature rise, at 40°C maximum ambient air temperature

Medical-Grade Transformers

Medical equipment has to be extremely sensitive to deal with low patient signal levels, and also be highly precise to enable accurate medical diagnosis. Transformers that are considered to be medical grade need to satisfy a number of additional safety guidelines and rules beyond those for non-medical-grade transformers. They are designed to protect caregivers, patients, and the equipment used in their care, so medical-grade transformers are required to maintain strict adherence to the International Electrotechnical Commission (IEC) 60601 standard and other internationally recognized safety agency standards.

Medical-grade transformers use toroidal construction (**Photo 8**), which has the following advantages: low stray electric and magnetic fields; high efficiency, tight regulation factor (ε), some as low as 0.007 (0.7%); and Faraday shielding between primary and secondary windings.

The effectiveness of an electrostatic field shield is largely independent of the geometry of the conductive material. Solid foil is better than a layer of wire. Silver is better than copper, copper is better than aluminum, and so on.

However, fixed or low-frequency magnetic fields can penetrate the conductive shield completely. Electromagnetic shielding requires soft magnetic materials.

In the case of varying electromagnetic fields, the higher the frequency the better the material resists magnetic field penetration. As with electrostatic shielding, electromagnetic shielding also depends on the electrical conductivity as well as the thickness of the shield.

Medical power isolation transformers (**Photo 9**) use medical-grade toroidal transformers installed in an enclosure. They are designed for a wide frequency range of 47Hz to 63Hz, with output voltage taps of 90, 100, 120, 208, 220, and 240 Vrms, to accommodate both 50Hz and 60Hz power sources.

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They are designed to certain standards and have several protective characteristics including:

- Low leakage current to ground, to equipment enclosures, and to the patient
- Low patient auxiliary current (measured between any single patient connection and all other patient connections of the same module or function, such as electrocardiogram electrodes)

Photo 8: Medical-grade toroidal transformers use toroidal construction. (Image Source: Triad Magnetics)

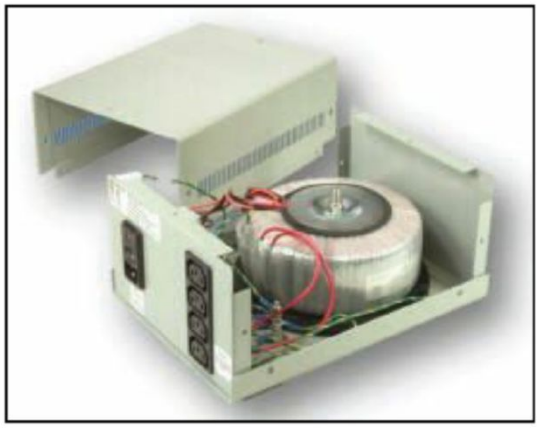


Photo 9: A medical power isolation transformer assembly, shown here, uses medical-grade toroidal transformers installed in an enclosure. (Image Source: Standex)

Photo 10: This is a MIL-PRF-27F Grade 5 encapsulated PC board-mounted power transformer. This particular one is a Pico Electronics 88660, 15VA toroidal transformer designed to MIL-PRF-27F Grade 5 Class S, with PC board pins (Image Source: Pico Electronics, Inc.)

- Continuous noise filtering, enhanced common mode surge protection, filters out EMI/RFI noise, utility switching transients, load-generated harmonics and ground loops
- Residual-current circuit breaker (RCCB) or ground fault circuit interrupter GFCI), which directly detect leakage current
- Built-in thermal protection for overload and short circuit conditions
- NEMA 5-20R hospital-grade output receptacles
- Built in accordance with ISO 9001:2015 certification
- UL/IEC 60601-1 compliant medical-grade listing
- Leakage current less than 100µA
- Double-reinforced insulation to withstand 5,000 Vrms Hi-Pot
- Efficiency: 90%-98%
- Low magnetic stray field
- Low mechanical noise
- Minimums on creepage distance and air clearance
- Cool operation with a maximum temperature rise between 25ºC to 55ºC.

There are a couple different types of medical-grade transformers. Safety ground transformers utilize standard transformer isolation. A safety shield between the primary and the secondary coils, with insulation between the shield and each coil, limits the maximum leakage current. If the isolation fails, the AC input current is immediately redirected to ground through the shield, which will trip the overcurrent device. The other type are double or reinforced insulation transformers, which can be larger than standard transformers, have two or more means of insulation. Should one insulation layer fail, the others will maintain isolation. If this second insulation layer fails, the overcurrent protection device will trip (fail safe).

Military-Grade Transformers

Military transformers are designed in accordance with MIL-PRF-27F (**Photo 10**). They have specific construction grades, and are designed to operate over the wide military maximum ambient temperatures, temperature rises, and electrical overloads. The 115V AC 400Hz power is used in aircraft, Navy submarines, space programs (Bendix made the 400Hz static inverters for the Gemini program) and even the AC power for some computer server rooms.

The transformers are available in two different 400Hz primary voltages, and with numerous secondary voltages from 5V AC to 115V AC. The 115V AC transformers are used to provide various

lower voltages, such as 26V AC, 36V AC, and 28V DC by means of transformer-rectifier units or switching power supplies. The 26V and 36V AC transformers are used to supply essential AC power for inertial navigation systems, autopilot, cockpit instrumentation, and low voltage lighting.

The Pico Electronics 88660 is fully specified. The no-load exciting current is 20mA. The primary and secondary DC resistances are 27.4Ω and 4.15Ω. The open circuit primary and secondary inductances at 400Hz are 694mH and 78.2mG. Typical efficiency and load regulation are 90% and 7%, respectively.

They also measured some typical winding parameters for us. For instance, the primary leakage inductance (400Hz secondary shorted) is 1.83mH. The secondary leakage inductance (400Hz primary shorted) is 216µH. The primary-secondary capacitance is 473pF. The class is identified by a single letter in accordance with MIL-PRF-27 as shown in **Table 1**, and denotes the maximum operating temperature of temperature rise plus maximum ambient temperature, and overloads of 12% for 48 hours.

For example, Class R has a maximum operating temperature designation of 105°C at sea level; this is normally at 65°C ambient plus a 40°C rise. If the temperature rise was determined to be 30°C, this same transformer could operate in an ambient as high as 75°C (75°C +30°C = 105°C). At higher altitudes, the temperature rise will increase and should be compensated for in the equipment design.

It is not recommended that a higher operating ambient temperature be specified than that to which the transformer or inductor will actually be exposed. To do so may result in a larger and heavier unit than is needed (violating SWaP-c). Transformers and inductors designed in accordance with this specification are capable of withstanding an overload of 12% for 48 hours.

Military-grade transformers are also subjected by the manufacturer to a barrage of MIL-PRF-27 qualification tests and standardization, most of which commercial transformers do not have to endure.

Once qualified, the transformer is assigned a MIL-PRF-27 qualified parts list (QPL) number and does not have to undergo any further quality testing unless there is a change in the design's form, fit, and/or function (FFF).

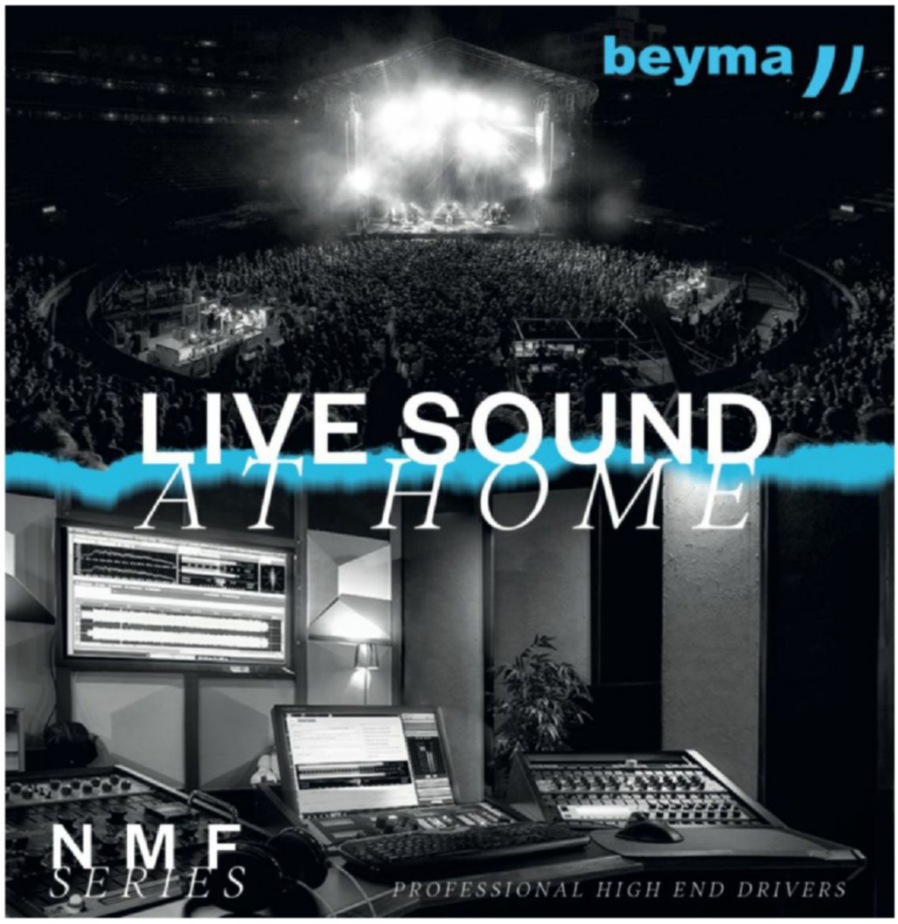
Transformer Coils

As transformer coils are being wound, the difference in voltage between adjacent turns in the same layer is quite small. However, the voltage difference between the start turn and the finish turn in an adjacent layer above or below it can be significant. This determines the thickness of the interwinding insulation required to prevent a flash-over failure.

Symbol	Maximum Operating Temperature
Q	85°C
R	105°C
S	130°C
V	155°C
T	170°C
U	170°C, as specified

Table 1: The class of military-grade transformer is identified by a single letter in accordance with MIL-PRF-27 and denotes the maximum operating temperature of temperature rise plus maximum ambient temperature, and overloads of 12% for 48 hours.

Commercial E-I transformers use cellulose impregnated paper to insulate the windings from the transformer core and metal frame, and the individual windings layers from each other. There are also standard-sized nylon bobbins, which completely separate the primary and secondary windings with barrier walls that save a lot of winding and construction time. They also reduce the capacitance and mutual inductance between windings, and allow for a higher dielectric breakdown voltage.



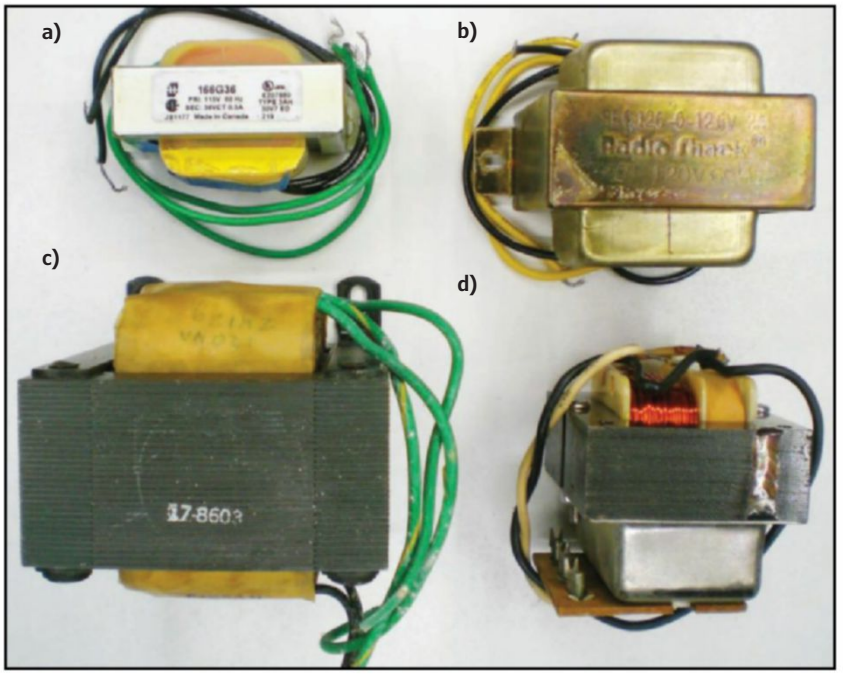
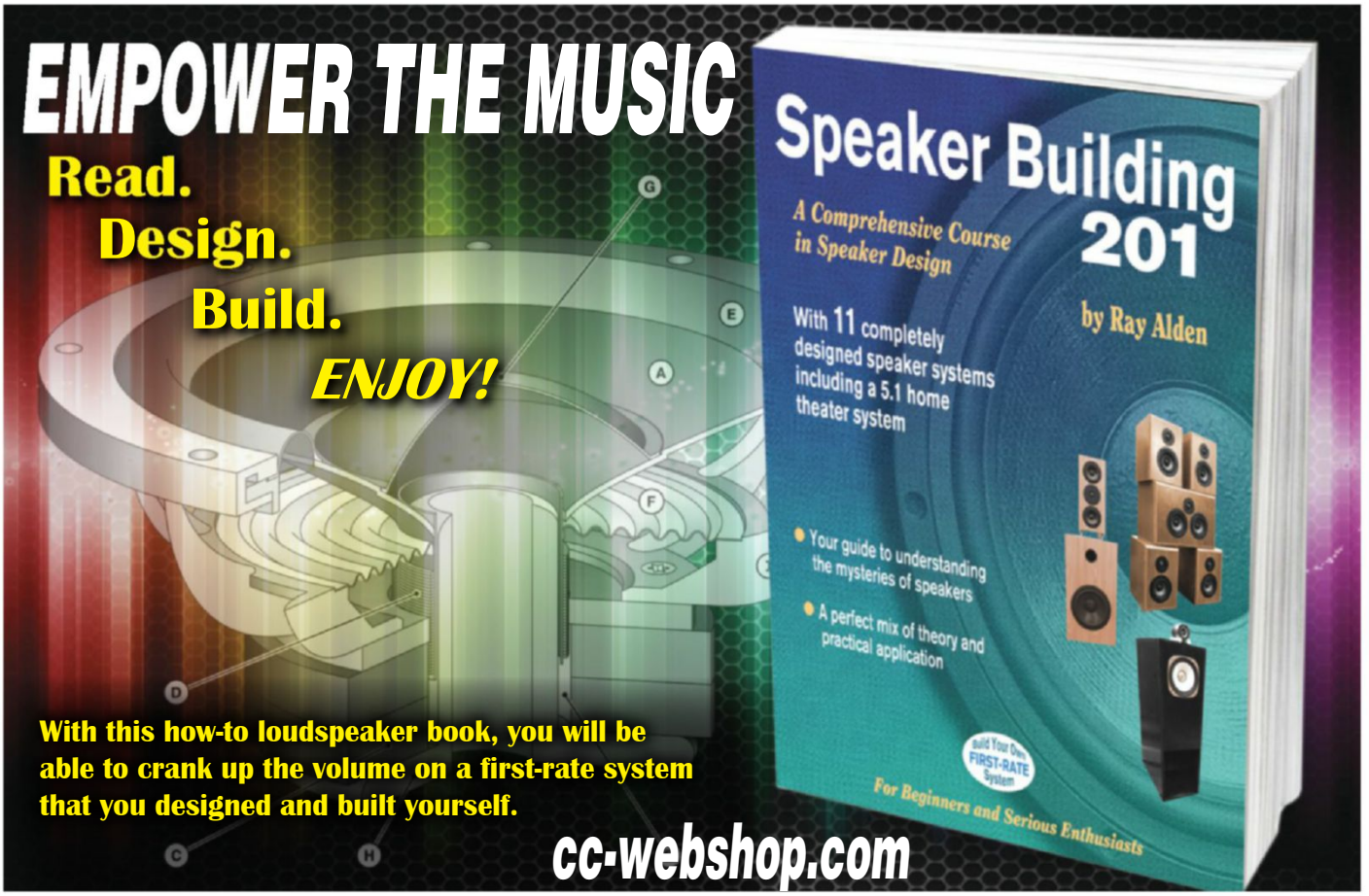


Photo 11: Layer-wound and bobbin-wound E-I core transformers. Left to right on top: (a) a Hammond 166G36 transformer; (b) a RadioShack RS 273-1512B transformer; left bottom (c) a Triad F-264U transformer; and lower right (d) an old doorbell transformer with the top end bell removed.

Photo 11 shows a group of E-I lamination transformers, left to right on top; (a) a Hammond 166G36 transformer rated 18 VA with a steel frame and nylon bobbin; (b) a RadioShack RS 273-1512B transformer rated 50.4 VA. It has a steel case with steel end bells for electromagnetic shielding, and a nylon bobbin.

On the left bottom (c) is a Triad F-264U transformer rated 120 VA. The laminations are exposed all around ("bare"), and the four mounting brackets hold the assembly together. The coils are layer wound with the primary on the bottom. Layer winding can result in a higher capacitance between the primary and secondary than bobbin-wound transformers.

I also took apart an old doorbell transformer (d), shown with the top end bell removed, and found it interesting that all the E laminations were stacked on the left side of the core, then the bobbin-wound coil was placed on the middle leg. Finally, all the I laminations were stacked on the other end, and the mating edges of the I and E laminations were welded together. That has to increase the core losses and reduce the efficiency. The terminal board on the outside of the lower end bell is for the low-voltage secondary connections.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MIL-Spec transformers use Dupont Kapton and glass-epoxy impregnated tape for electrical insulation. All insulating materials are controlled by MILI24768. For C and E laminated cores, bobbins or rectangular core tubes are pre-wound with heavy ML wire, placed over the legs of the core, then the core halves are mated and held together with a 316 stainless steel band using custom stainless-steel retaining clips.

Photo 12 shows four typical C and E core tube, or core box, materials. The largest one on the left (a) as well as the third from left (c) are MIL-I-24768 type GEBF glass-filled epoxy resin, high-temperature arc and flash resistant insulations. The typical industry identification is G10.

The second from left, (b) in Photo 12, uses a special nomex-kapton-nomex construction. Finally, the core tube on the right (d) is M24768 type PBG, paper-based phenolic-resin, general purpose. PBG is generally used for prototype designs or custom lab transformers that will not be undergoing environmental, vibration, or shock testing. The rectangular core tubes are used for shell-type transformers using two C cores with a single winding in the center. Square core tubes are used for small transformers that use just a single C core. Custom bobbins and terminal blocks can be made using additive 3D manufacturing with high-temperature insulating polymers.

Some commercial E-I core transformers use a cellulose paper that has outstanding electrical insulation properties. It is applied in layers and extends slightly past the copper end turns of each winding (see the arrow in **Photo 13**). Large utility power transformers are oil-cooled and use oil-impregnated cellulose insulation.

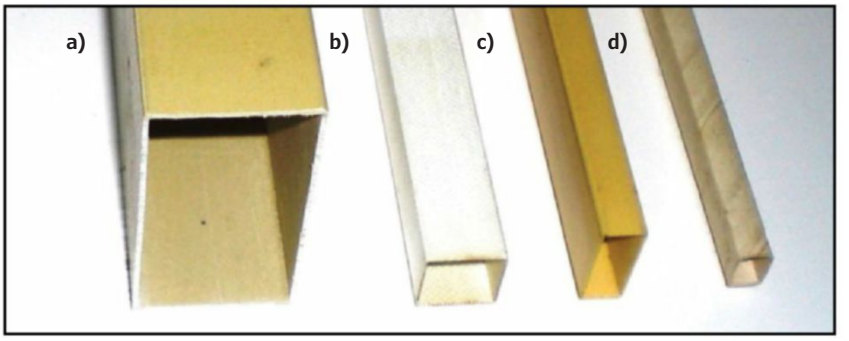

Photo 12: These are typical C and E core tube, or core box, materials. The largest one on the left (a) as well as the third from left (c) are MILI24768-type GEBF insulations. The second from left (b) uses a special nomex-kapton-nomex construction, and the core tube on the right (d) is M24768 type PBG.

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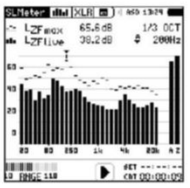
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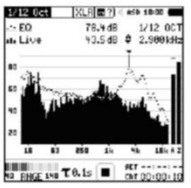
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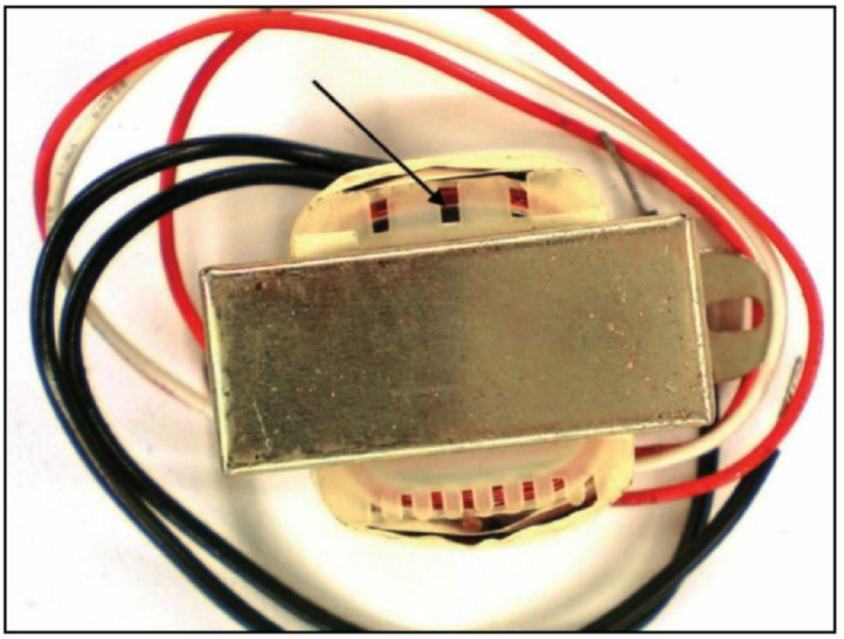

Photo 13: Cellulose paper insulation between windings (arrow) is shown on a nylon bobbin transformer.

Newer commercial E-I core transformers use dual-section nylon or polypropylene bobbins that provide excellent dielectric isolation between the primary and secondary windings, and between the windings and the core (**Photo 14**).

The number of volts-per-turn of wire (V/N) is proportional to the lowest AC frequency (f), the peak continuous induction (B) we design for, and the effective core cross-sectional area (A). B and f are usually fixed values, leaving the core area as a trade-off. This depends on the core areas of available standard cores. The optimum core area per unit weight for C and E core transformers is a square, where the steel strip width is equal to the effective metallic buildup height of the core laminations.

The required core weight is proportional to the required output power in watts divided by the lowest AC frequency (f), using the effective core cross-section and the magnetic length of the core as the variables used to select a core weight as close as possible to the calculated required weight. Core catalogs publish the core weights, and watts per pound for each of their standard core part numbers.

There are times in aerospace when we will use a custom core to achieve the desired SwaP-c. There

is usually a non-recurring cost involved to tool up for a custom core.

Magnet Wire and Connection Leads

Magnet wire for commercial transformers and inductors is usually coated with one or more thin layers of a polyvinyl formal acetate. Polyvinyl insulation is rated for operation from -40°C to 105°C. The insulated connecting leads are usually thermoplastic PVC.

Aerospace magnet wire is coated with one or more thin layers of polyimide resin, which can operate at temperatures up to +260°C. It is available in round, square, and rectangular cross-sections. The silver-plated stranded copper connecting leads are insulated with MIL-W-16878/4 Type E Dupont Teflon (PTFE).

The voltage increase between each turn of wire in a layer depends on the calculated volts-per-turn. At the ends of each layer is a cumulative increase in voltage to adjacent layers depending on the mean magnetic path length in the chosen core and the number of layers of wire. The inter-layer insulation has to account for this. Three-phase 115V AC 400Hz aerospace power transformers can be made with a secondary voltage as high as 750V AC. Of course, in consumer and commercial audio applications, the use of materials and processes found in aerospace applications would probably be cost prohibitive.

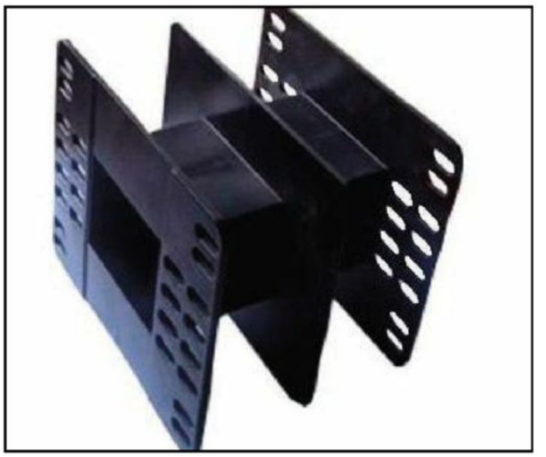

Photo 14: Newer commercial E-I core transformers use dual-section nylon or polypropylene bobbins. Shown here is a tranformer bobbin T08. (Image Source: Bhavna Plastics)

Peak Operating Voltage

Operating voltage, sometimes referred to as working voltage, is defined in standards such as Underwriters Laboratories UL 60950. It is the highest voltage to which the insulation or the component under consideration is, or can be,

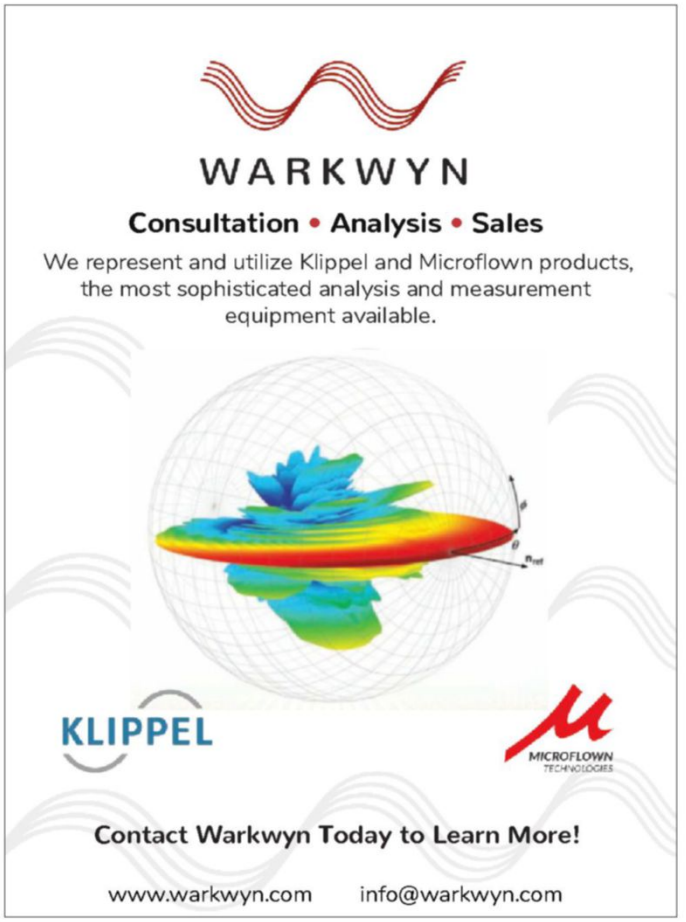
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Photo 15: After transformer construction is finished, a high-potential (hi-pot) voltage test is applied with special insulation resistance and high-voltage test equipment. Shown here is a Bendix Type 60B4-1-A AC/DC Insulation and hi-pot tester.

subjected when the equipment is operating under conditions of normal use.

Steady-state voltages above 42.4V peak or 60V DC, in computers and similar office equipment, are generally considered hazardous.

High-Potential Testing

After transformer construction is finished, a high-potential (hi-pot) voltage test is applied with special insulation resistance and high-voltage test equipment (**Photo 15**) that limits the current and disconnects the high voltage in case of a hi-pot failure.

This minimizes subsequent arc damage and allows for possible winding repairs. The test is performed with all the windings shorted together, with the test voltage applied from windings to metal chassis or case ground. Hi-pot between secondary and primary windings is applied with the wire end turns of each winding individually shorted to each other.

The hi-pot requirement for all commercial transformers with 120V AC or 240V AC primary windings is 1,250V AC to 1,500V AC. The commercial specs allow for a DC hi-pot voltage, but it must be increased by a factor of √2 times (1.414) to equal the AC peak voltage, and applied in each polarity. Insulation resistance tests are performed with at least 500V DC.

The UL/CSA/CE conformity marks are required to sell all household consumer equipment in the US, Canada, and Europe. UL 1585 determines the dielectric voltage-withstanding test for Class 2 and Class 3 transformers.

Earlier commercial 115V AC transformer designs with secondary windings of 50V AC or less could place the secondary winding closest to the core since they could be hi-pot tested at the lower 600 +50/0V AC maximum limit for 12 seconds allowed at that time. This compromised the primary winding somewhat in that it had higher resistance than if it was placed closest to the transformer core.

The MIL-Spec hi-pot limit is 60Hz, 1,000V AC plus twice the RMS operating voltage. The test current limit level has to be adjusted for the capacitive coupling between the windings. There is no DC hi-pot test specified or required. The insulation resistance test voltage is at least 600V DC.

Next Month

Part 4 of this series will discuss the various parameters of transformers and their losses.

About the Author

Chuck Hansen is an Electrical Engineer and holds five patents in his field of electric power engineering. He has worked in both the electric utility and aerospace electric power industries. He was assigned as the Instrument and Controls Startup Group Lead Engineer for two nuclear power generating units. He was the Electrical Systems and Controls Supervisor for a number of military programs as well as commercial and business jet electric power programs. He has written two books for Audio Amateur publications, and more than 260 magazine articles. He began building vacuum-tube audio equipment in college, and enjoys restoring and modifying audio equipment and test equipment. In his most recent roles as Senior Design Engineer and Lab Calibration Manager, he designed specialized test equipment particular to the aerospace electric power sector.

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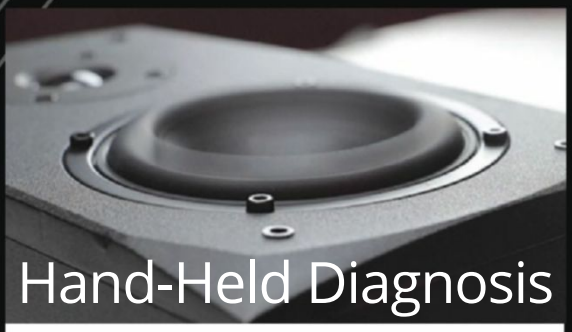

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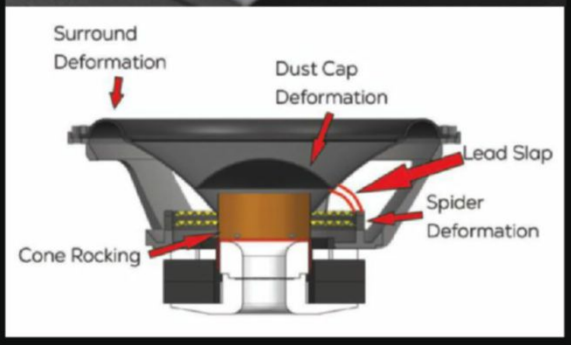

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